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[54] **SYNCHRONIZATION AND SAMPLING FREQUENCY IN AN APPARATUS RECEIVING OFDM MODULATED TRANSMISSIONS**

[75] Inventors: **Jari Junell**, Vantaa; **Mika Kasslin**; **Mikko Kokkonen**, both of Helsinki, all of Finland

[73] Assignee: **Nokia Technology GmbH**, Pforzheim, Germany

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[51] **Int. Cl.**[7] **H04J 11/00**; H04J 1/00

[52] **U.S. Cl.** **370/503**; 370/208; 375/324; 375/364

[58] **Field of Search** 370/206, 208, 370/210, 503, 509, 512, 513, 514, 516, 517; 375/203, 344, 340, 342, 354, 362, 363, 364, 355

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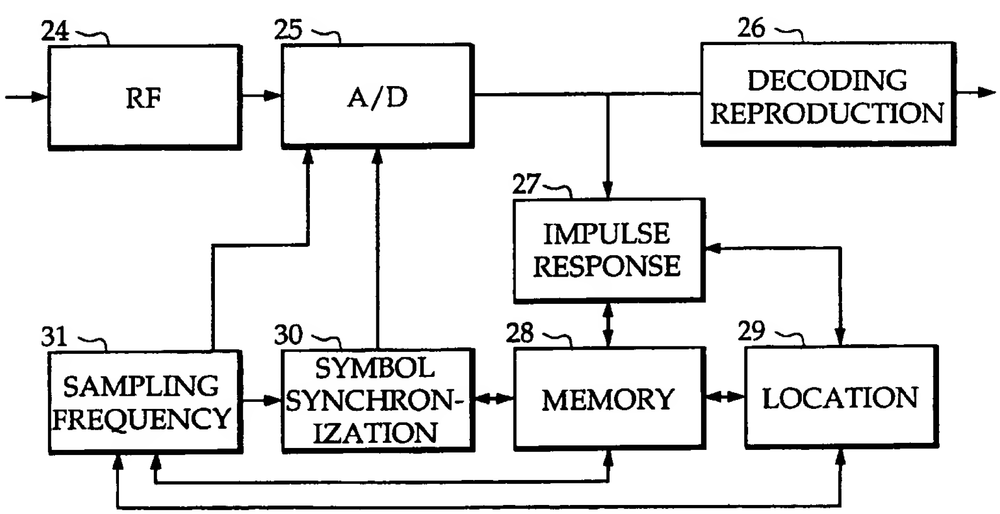
Primary Examiner—Chi H. Pham
Assistant Examiner—Steven Nguyen
Attorney, Agent, or Firm—Ware, Fressola, Van Der Sluys & Adolphson LLP

[57] **ABSTRACT**

An OFDM receiver determines (**12**; **27**) the pulse response of a radio channel and locates (**13**; **14**) its starting point, end point and the maximum and its value. The difference between the end point and the starting point gives the length of the pulse response. A guard interval time corresponding to the guard interval separating the OFDM symbols is set (**17**; **18**) in the receiver in such a manner that it covers the most significant components of the pulse response. A slow and monotonous temporal shift of the pulse response between measurement rounds indicates an error in the sampling frequency. The error is corrected (**23**; **31**) in such a manner that the pulse response shift is compensated for.

11 Claims, 4 Drawing Sheets



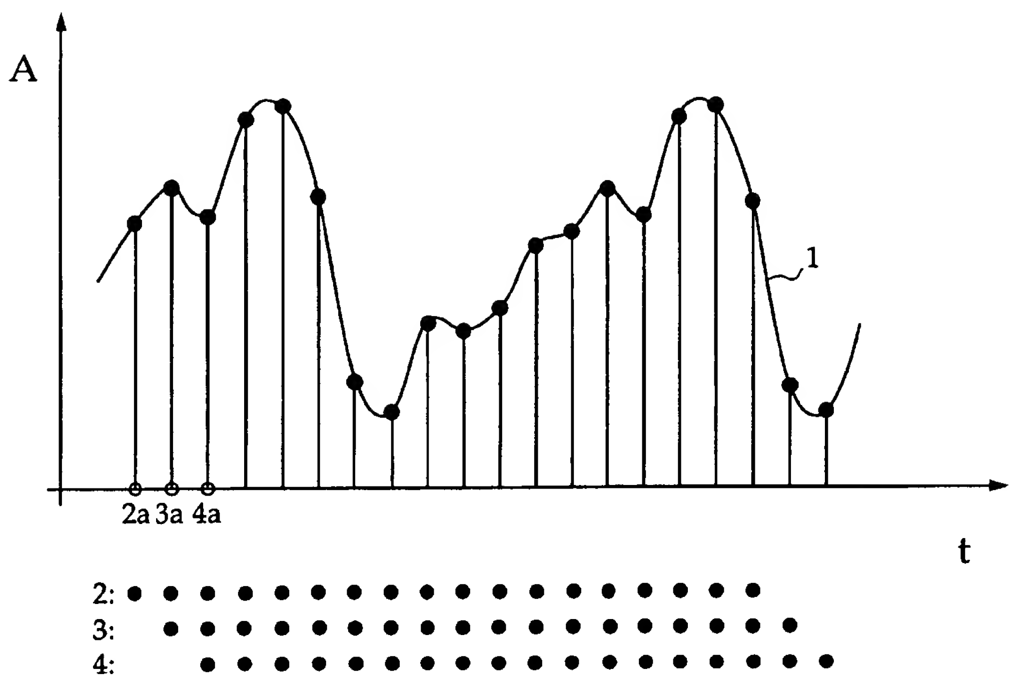


FIG. 1
PRIOR ART

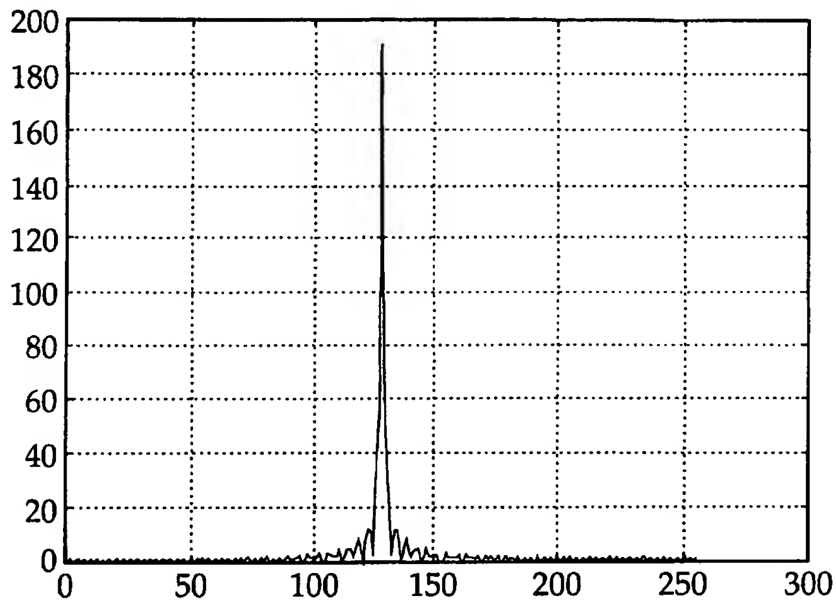


FIG. 2
PRIOR ART

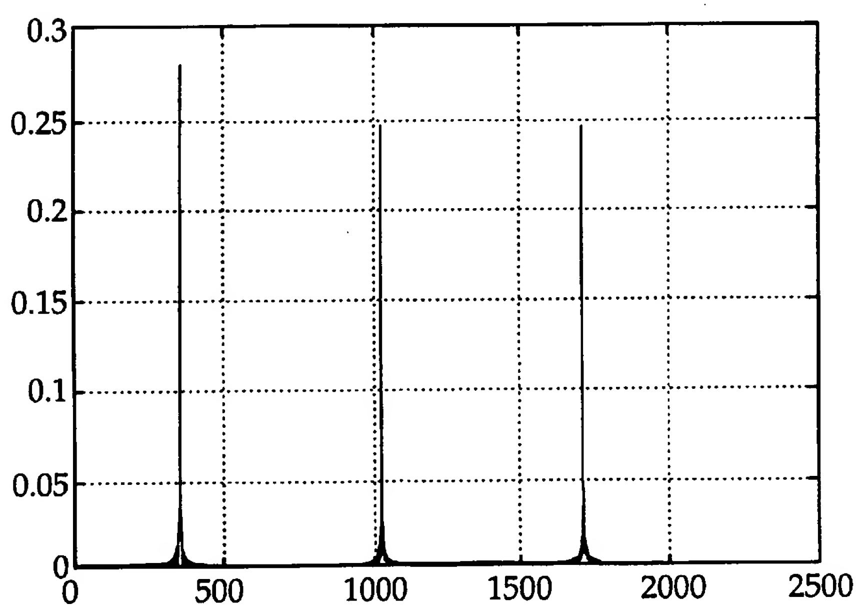


FIG. 3

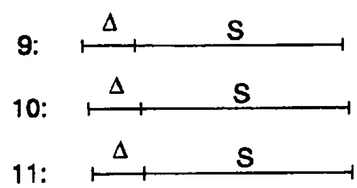


Fig. 4a

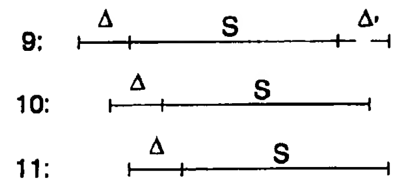


Fig. 4b

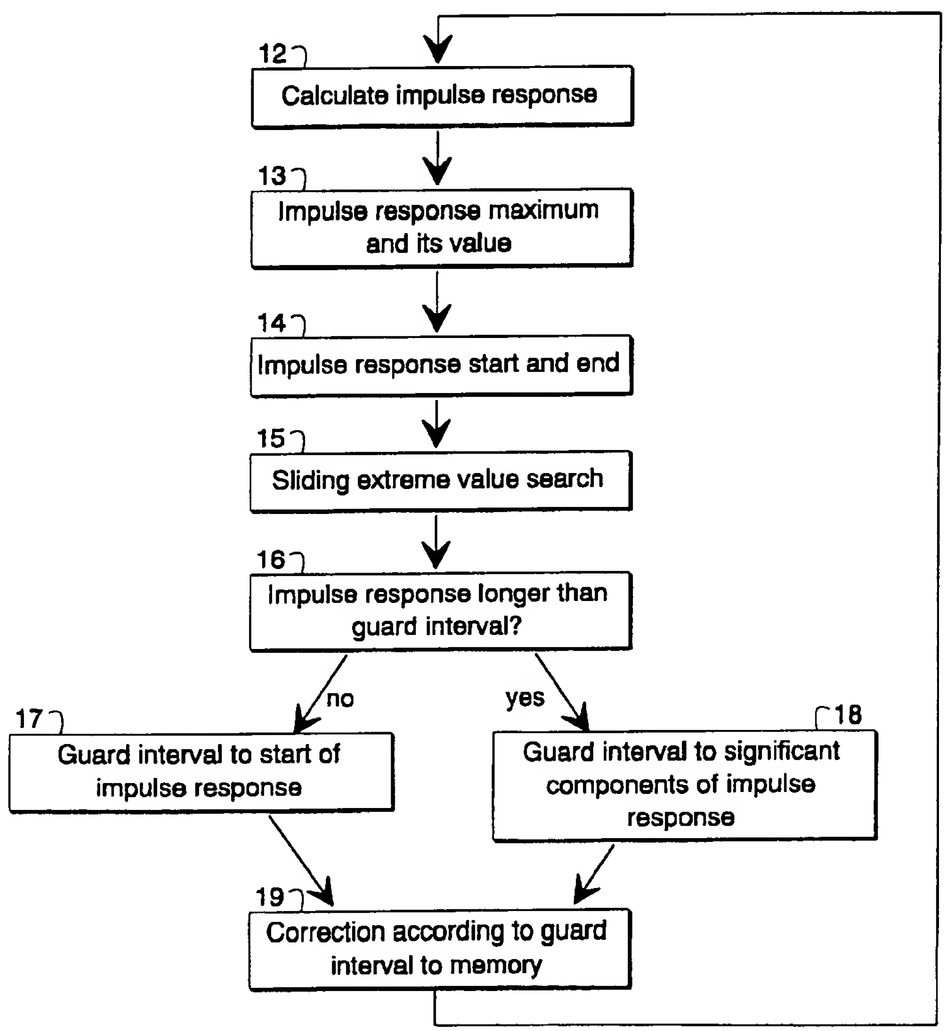


Fig. 5

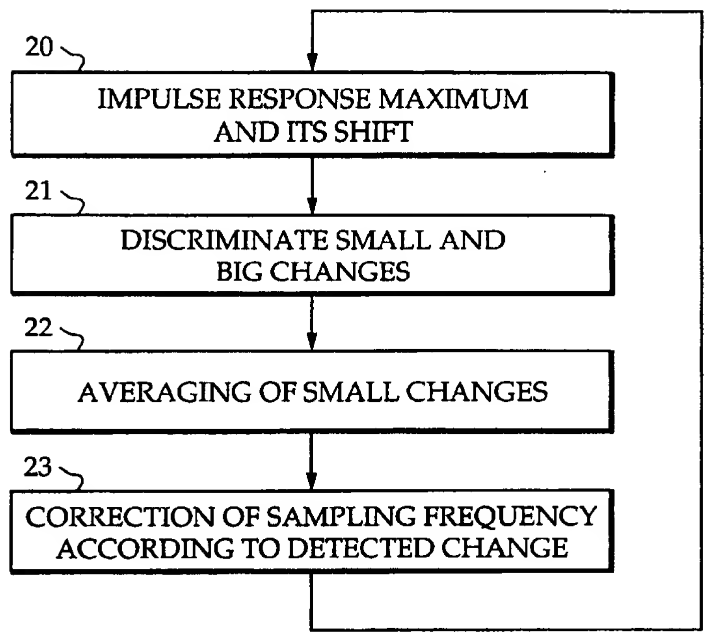

20
IMPULSE RESPONSE MAXIMUM AND ITS SHIFT
21
DISCRIMINATE SMALL AND BIG CHANGES
22
AVERAGING OF SMALL CHANGES
23
CORRECTION OF SAMPLING FREQUENCY ACCORDING TO DETECTED CHANGE

FIG. 6

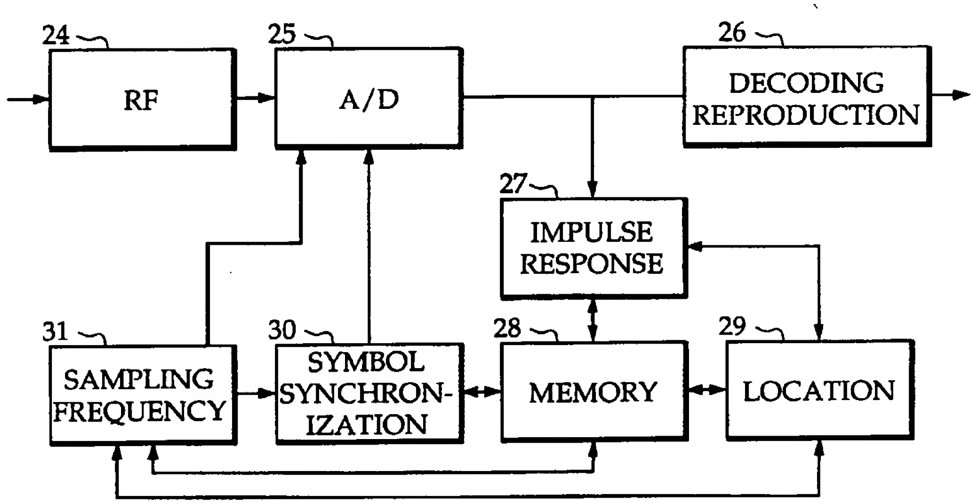

24
RF
25
A/D
26
DECODING REPRODUCTION
27
IMPULSE RESPONSE
31
SAMPLING FREQUENCY
30
SYMBOL SYNCHRON-IZATION
28
MEMORY
29
LOCATION

FIG. 7

SYNCHRONIZATION AND SAMPLING FREQUENCY IN AN APPARATUS RECEIVING OFDM MODULATED TRANSMISSIONS

BACKGROUND OF THE INVENTION

1. Technical Field

The invention relates in general to the adjustment of reception characteristics in an apparatus receiving radio-frequency transmissions and in particular to the automatic adjustment of timing and sampling frequency in an apparatus that receives OFDM modulated transmission.

2. Discussion of Related Art

Orthogonal frequency division multiplex (OFDM) refers to a modulation method where the transmitting device divides and attaches the transmitted signal to several subcarriers which are located on the frequency axis at regular intervals on a certain frequency band and which are sent simultaneously. Known radio-frequency communication systems that employ OFDM modulation include the DAB (Digital Audio Broadcasting) and DVB (Digital Video Broadcasting) systems. The former is specified in general outline in the ETS 300 401 standard by the European Broadcasting Union (EBU) and the European Telecommunications Standards Institute (ETSI), and the latter is specified in general outline in the prETS 300 800 draft standard by the same organizations. In these systems, a section of a digital signal to be transmitted on a certain subcarrier is encoded into phase and/or amplitude changes with respect to a certain known phase. That time slice of the transmitted signal during which the modulating phase state is constant separately at each subcarrier frequency is called a OFDM symbol, or a symbol in short.

In order for the receiving device to be able to correctly interpret the phase changes on the different subcarriers, the transmitter must include a certain phase reference in the signal. In the DAB system, the transmitted signal is divided into 24-ms or 96-ms frames, depending on the transmission mode, and each frame has a phase reference symbol at the beginning (after the null symbol) which indicates the phase reference simultaneously to all subcarriers. In the DVB system, the phase reference is included in the so-called pilot channels which are found in each symbol at intervals of twelve subcarriers.

Successful OFDM reception requires that the receiver maintains the correct symbol synchronization and sampling frequency. Symbol synchronization means that the receiver knows at which point of time each symbol begins and times the symbol detection correspondingly. Sampling frequency refers here to the frequency at which the A/D converter in the receiver takes samples from the received analog oscillation in order to convert the signal into digital form, whereby the A/D converter and subsequent circuits can interpret to which bits or bit combinations in the digital data flow the signal phase changes refer. In addition, the receiver has to maintain frequency synchronization, i.e. to tune the reception and mixing circuits so that the detected frequency band covers all subcarriers of the OFDM signal at an accuracy which is less than half of the difference between two adjacent subcarriers. Maintaining the symbol synchronization, sampling frequency and frequency synchronization is especially difficult if the transmitter and receiver are moving with respect to each other. The receiver may be located in a car, for example, and as the car moves around in an urban environment, the propagation path of the radio signal changes constantly, resulting in attenuation and

reflections. The receiver may also be located in a satellite, and as the satellite moves, the speed difference between the receiver and the satellite changes, being possibly up to several kilometers per second.

A method is known from Finnish Patent Applications No. 962138 "Vastaanottimen tahdistuminen joutotilassa" and No. 962139 "Signaalin haku eräässä satelliittipuhe-linjärjestelmässä" to produce and maintain symbol synchronization and frequency synchronization in a radio system that does not use OFDM modulation. The method is based on the fact that a received signal includes on a certain control channel a synchronization sequence comprising bits in succession and a frequency information part which contains a short duration of pure sine wave at a desired frequency. The synchronization sequence belongs as part of the rest of the signal to a higher-power burst, and the receiver gets the coarse frame synchronization just by monitoring the highest received power peaks. In the finer synchronization, the receiver calculates how the various timing errors affect the detection of the synchronization sequence and deduces how the timing of the sampling should be corrected to make the received synchronization sequence match better with the known format of the synchronization sequence. Frequency fine-tuning is performed by calculating for a discrete Fourier transform from the received frequency information part and by tuning the reception and mixing frequencies so that the peak of the frequency error spectrum yielded by the Fourier transform is as close to zero as possible. In addition, the receiver monitors how the timing and frequency parameters change and predicts from them the required corrections while in idle state, i.e. receiving only occasionally.

The prior art method described above is not suitable to be used as the synchronization method for an OFDM receiver since an OFDM modulated signal does not include separate synchronization sequences or frequency information parts like the control channel signal of the I-CO Global Communications system described in said patent applications. There exists no efficient prior art method for maintaining the symbol synchronization, sampling frequency and frequency synchronization in an OFDM receiver.

SUMMARY OF INVENTION

An object of the invention is to provide a method and an apparatus for adjusting the symbol synchronization and sampling frequency in an apparatus receiving OFDM modulated transmissions. A particular object of the invention is that the method according to the invention will not require an unreasonably high computing capacity or special components that are difficult to produce, so that the apparatus implementing the method be suitable for large-scale mass production.

The objects of the invention are achieved by using known parts of the received signal to calculate an instantaneous pulse response for the radio channel and by comparing changes in the pulse response with pulse responses calculated earlier and by correcting the synchronization and sampling frequency in order to compensate for the changes in the pulse response.

The method according to the invention is characterized in that it comprises steps wherein

- a response is determined for the radio channel on which the receiver is receiving the OFDM signal in question, and
- the guard interval time is set in the reception such that it coincides with the most significant components of the pulse response corresponding to one symbol.

The invention is also directed to a receiver apparatus which is characterized in that it comprises

means for determining the instantaneous pulse response for the radio channel,

a synchronizable A/D conversion circuit the operation of which includes cyclically repeated guard interval time and information time, and

means for driving said A/D conversion circuit to a state in which said guard interval time covers the period of time of the calculated pulse response that includes the most significant components of the pulse response.

The invention is based on the utilization of time-domain correlation characteristics of the reference signal in an OFDM transmission. In the DAB system, the reference signal means a phase reference symbol, and cross-correlation between the received format and the known format of that symbol yields the instantaneous pulse response. In the DVB system, the pulse response is estimated from scattered pilot subcarriers for four consecutive symbols. The required changes in the symbol synchronization and sampling frequency can be deduced by monitoring how the pulse response changes from a measurement to another. The symbol synchronization is preferably set so that the guard interval between the symbols coincides with the beginning of the correlation function representing the pulse response.

A sampling frequency error shows as a slow and monotonously continuous shift of the maximum of the correlation function representing the pulse response. By correcting the sampling frequency the receiver attempts to eliminate said change.

BRIEF DESCRIPTION OF THE DRAWING

The invention is described in more detail with reference to the preferred embodiments, presented by way of example, and to the attached drawing, where

FIG. **1** shows how a certain signal is sampled in a known manner,

FIG. **2** shows a known absolute value of the auto-correlation function of the phase reference symbol in the DAB system,

FIG. **3** shows a known absolute value of the auto-correlation function determined from the pilot subcarriers in the DVB system,

FIGS. **4***a* and **4***b* show certain timing situations in the symbol reception,

FIG. **5** shows the flow chart of the method according to the invention for correcting the symbol synchronization,

FIG. **6** shows the flow chart of the method according to the invention for correcting the sampling frequency, and

FIG. **7** shows the block diagram of the receiver according to the invention.

Determining the radio channel's pulse response in the receiver is a procedure known in the prior art. As far as the invention is concerned, it is in fact irrelevant how the pulse response is determined, but in order to provide sufficient background information for the invention we will below describe one illustrative method for calculating the pulse response. This method for calculating the pulse response is based on the temporal cross-correlation of a certain received signal part and its known format. Temporal cross-correlation of two signals generally refers to the accuracy with which the signals yield the same values at certain moments of comparison. The correlation can be calculated using a known algorithm.

FIG. **1** shows how a certain signal is sampled in a known manner for calculating the cross-correlation. The horizontal axis in the figure represents time t and the vertical axis schematically depicts the signal's amplitude A. Curve **1** represents a certain part of the received signal, and the receiver has prior knowledge as to the supposed format of said signal part. An A/D converter in the receiver takes samples of the signal at regular intervals, described by vertical lines in the figure. The first sample sequence **2** comprises the samples that begin from a certain first sampling location **2***a* and are distributed at regular intervals over a time span which corresponds to the temporal duration of the known signal. As the receiver cannot be sure about the exact beginning of the known signal, it takes a second sample sequence **3**, which begins one sample later at location **3***a* and lasts one sample longer. The figure also shows a third sample sequence **4** which begins at location **4***a*. There can be as many sample sequences as the receiver is capable of processing with its memory and processing capacity.

The signal format known to the receiver is stored as samples in the receiver's memory. To calculate the cross-correlation the receiver multiplies, sample by sample, the sample sequence stored in the memory by a certain sample sequence obtained by sampling and summing the results. The more accurately the sample sequence corresponds to the known format of the signal, the higher the cross-correlation value. Signal auto-correlation refers to the result produced by the algorithm used for calculating the cross-correlation when the signal is compared to itself FIG. **2** shows the absolute value of the auto-correlation function of the phase reference symbol in the DAB system. Locations on the horizontal axis represent the beginnings of the sample sequences and the vertical axis represents the numerical value of the function. The figure shows that the sampling sequence the number of which is approximately 128 yields by far the highest correlation value, i.e. it has the best correspondence to the correct timing of the sampling.

In practice, it is not sensible to use the direct method described above to calculate cross-correlations that contain dozens of sample sequences to process but instead to use a method wherein the signal is sampled once and a complex Fourier transform is performed on the sequence of samples, thus taking the problem from the time domain to the frequency domain. The receiver gives the complex frequency spectrum produced by the Fourier transform location by location on a reference spectrum which is a complex conjugate of the frequency spectrum of the known format of the signal. By inverse-transforming the obtained result we get directly the curve for the correlation function, which in the case of auto-correlation conforms to FIG. **2**.

FIG. **3** shows the absolute value of an auto-correlation function calculated from four consecutive symbols in the DVB system, wherein the signal part under examination consists of the sum of the pilot channels included in the symbols. The figure shows three successive peaks with a time difference of one-third of an effective symbol (by combining the scattered pilot channels of four symbols we get a symbol for which it is known every third carrier wave). In order for the description below and its algorithms to be similarly applicable both in the DAB and in the DVB systems, we will examine the pulse response curve of the DVB system in such a manner that we only take a certain band around the highest peak, which can be spread on the same horizontal axis scale as the DAB pulse response curve shown in FIG. **2**.

Next it will be discussed how changing the radio channel characteristics affects the pulse response curve shown in the manner according to FIGS. **2** and **3**. The maximum value of the pulse response curve at the peak location representing

the best correlation depends in principle on the signal path attenuation, i.e. the more the signal is attenuated on the way from the transmitter to the receiver, the lower the peak of the curve. In practice, the automatic gain control (AGC) circuit of the receiver evens out the effect of the changing attenuation in the case of an unequivocal peak. If the signal propagates from the transmitter to the receiver via several parallel propagation paths which produce different propagation delays, the curve shows, instead of one peak, several peaks close to each other. Then, their relative heights are significant, because the highest peak corresponds to the propagation path on which the signal is attenuated the least. If the timing used by the receiver is changed, i.e. the beginning of sampling is shifted with respect to the actual contents of the received signal, the peak representing the best correlation moves to the right or to the left on the horizontal axis.

Successive symbols in an OFDM modulated transmission are separated by so-called guard intervals that provide with their contents a characteristic useful from the point of view of the present invention: the interface between a guard interval and the symbol following it does not contain a phase discontinuity at any subcarrier frequency. So, regarding phase information, the contents of a guard interval are the same as those of the symbol following it. FIGS. 4*a* and 4*b* show two timing situations, wherein a certain symbol part S containing information and a guard interval A preceding it arrive in a receiver via three different routes, each of which produces a delay unequal to the others. In FIG. 4*a*, the delays are almost the same, and the symbol echoes propagating via routes **9**, **10** and **11** arrive in the receiver almost simultaneously. In FIG. 4*b*, the differences between the delays are considerably bigger, and so the symbol echoes arriving via different routes arrive at different times. If the calculation method for a radio channel pulse response described above is applied in these timing situations, the pulse response curve produced by FIG. 4*a* is a single peak, only a little widened, whereas the curve produced by FIG. 4*b* shows clearly three different peaks due to the fact that cross-correlation yields a relatively good result for each symbol echo arrived at a different time.

The main purpose of the symbol synchronization adjustment algorithm according to the invention is to maintain receiver timing in such a manner that the significant components of the pulse response fall within the period of time defined as the guard interval in the receiver. The foundation of this purpose can be found by examining FIGS. 4*a* and 4*b*. The receiver, which according to the invention makes the guard interval to begin from the moment at which the symbol echo that propagated via the fastest propagation path (in FIGS. 4*a* and 4*b*, the symbol of propagation path **9**) causes a first significant correlation peak, starts to read the information contents proper of the symbol from the location where the information part S of that same (fastest) symbol echo begins. Echoes of the same symbol arriving via other propagation paths (propagation paths **10** and **11**) may at that point still contain guard intervals. However, as the phase contents of the guard interval are the same as the phase contents of the symbol part containing information and there are no frequency hops between the guard interval A and the symbol part S, the timing will not cause phase information crosstalk between successive symbols. If the receiver timed its operation in such a way that its notion about the guard interval would coincide with, say, the middlemost received symbol echo (in FIGS. 4*a* and 4*b*, the symbol echo arriving via propagation path **10**), it would also time the symbol information contents read procedure in such a way that it would coincide with the information part of the middlemost echo. FIG. 4*b* shows that at the end of this period of time there already arrives via the fastest propagation path a guard interval Δ' which belongs to the next symbol and may have a different phase content, thus resulting in an error in the information part read by the receiver.

Formulated as a systematic algorithm, the symbol synchrononization method according to the invention is as shown in FIG. **5**. In step **12** the receiver calculates the radio channel pulse response using the method described above or a corresponding method. In step **13** it finds the pulse response maximum and stores its location and value in a memory. In step **14** the receiver finds the earliest and the latest significant components of the pulse response by reading the pulse response curve both from the beginning and from the end to the middle. The receiver stores the locations of points in which the value of the pulse response curve is for the first time a certain fraction of the stored maximum, as read both from the beginning and from the end. Said fraction can be, say, ⅛ or other threshold value found suitable by experimentation.

In step **15** the receiver performs a sliding extreme value search for two successive estimation rounds. This means that it compares the latest stored location of the first significant component of the pulse response with the location of the corresponding component stored during the previous estimation round and chooses the earlier of these two locations as the beginning of the pulse response. Similarly, the receiver compares the latest detected end of the pulse response with the end detected in the previous round and chooses the later of these values. At the same time, however, it saves the currently stored pulse response beginning and end for the comparison in the next round. The aim of said sliding extreme value search is to eliminate the effect of sudden disturbances. The receiver calculates the length of the pulse response by subtracting the location of the beginning from the location of the end. It is assumed here that the receiver knows how the scale of the horizontal axis of the graph of the pulse response corresponds to real time.

Step **17** is performed if it is detected in the inference step **16** that the length of the pulse response calculated by the receiver is shorter than the length of the guard interval between OFDM symbols which is known to the receiver. The receiver chooses the beginning of the pulse response determined in the previous step as the beginning of the guard interval to be used in the received symbol interpretation. If the calculation shows that the length of the pulse response is greater than the length of the guard interval, the receiver performs the next step **18**, wherein it sets the guard interval for the symbol interpretation such that the sum of the absolute values of the pulse response components left outside the guard interval is as small as possible. Finally, in step **19**, the receiver corrects the pulse response beginning and end estimates in its memory so that they correspond to the correction of guard interval location performed in step **17** or **18**. The algorithm then returns to its starting point.

To adjust the sampling frequency the method according to the invention includes a classification step, in which the receiver examines how big changes there have occurred in the pulse response timing. Big changes are caused by radical changes in the radio channel, e.g. when new propagation paths appear or old ones are lost.

Small but repeated changes in the same direction are caused by the fact that the sampling frequency of the receiver is not exactly as it should be. If the sampling frequency of the receiver is too high, the receiver will take

an amount of samples corresponding to one symbol from a period of time which is shorter than a true symbol. The receiver starts taking samples for the next symbol too early and the pulse response maximum shifts to a later point. Similarly, if the receiver's sampling frequency is too low, it will not have time to take an amount of samples corresponding to one symbol before the next symbol starts and the pulse response maximum shifts to an earlier point.

FIG. **6** shows a flow chart of the algorithm for that part of the invention which adjusts the sampling frequency. In step **20** the receiver finds the location for the pulse response maximum (or reads it from memory if it was stored in step **13** of FIG. **5**) and calculates how much the maximum has moved with respect to the maximum of the previous estimation round. The difference between the maximum locations is stored in memory. In step **21** the receiver sorts nine successive differences according to their magnitude and selects the three middlemost ones. This procedure corresponds in a way to low pass filtering since therein the receiver assumes that the three biggest differences, in absolute values, at both extremes (positive and negative directions) correspond to the aforementioned "big changes", i.e. are caused by other reasons than a sampling frequency error. Here, we have chosen the figures three and nine by experimentation, and the invention does not in fact place any restriction as to how the examination is limited to small enough changes. In step **22** the receiver calculates the average ("ave") of the differences it has selected. The relation of parameter "ave" to the time difference between the moments of estimation for two pulse responses yields a ratio which gives the correction to the sampling interval. Mathematically presented, the correction ΔT to the sampling interval length T is obtained as follows:

$$\Delta T = \frac{\text{ave}}{\text{frame\_length}} T,$$

where parameter frame_length equals the time difference between the estimation moments of two pulse responses. Its unit must be the same as that of parameter "ave". The parameters are preferably expressed as multiples of the sampling interval. In step **23** of FIG. **6** the receiver calculates the correction to the sampling interval according to the equation above. The new sampling frequency equals the inverse value of the new sampling interval.

The name of parameter frame_length suggests that in a frame-based system, such as the DAB, the pulse response estimation and the sampling frequency correction following it are preferably performed at intervals of one frame because there is at the beginning of each frame a phase reference symbol. In systems like the DVB, where information is not transmitted in frames, the length of parameter frame_length can be freely chosen and it may even change according to the operating mode of the receiver. According to an advantageous embodiment, a DVB receiver checks whether it gets its operating voltage from a fixed electric network or from a portable power source. A fixed electric network means that the receiver is probably unmoving and the characteristics of the radio channel will change only slightly and, therefore, the pulse response estimation is not needed very often. A DVB receiver with a portable power source may be moving and, therefore, the pulse response estimation and the symbol synchronization and sampling frequency corrections following it should be performed more often.

In DVB and other systems based on coherent detection, the symbol synchronization and/or sampling frequency correction has an effect on the channel estimate as well, so the receiver has to take this change into account and compensate for it by adjusting the channel tuning. In systems like the DAB, there is at the beginning of each frame a phase reference after which the detection of the same frame is performed as differential detection. This kind of reception arrangement automatically takes into account the channel estimate correction, as long as the corrections are made at the frame boundary.

FIG. **7** shows schematically a digital OFDM receiver which can be applied to implement the method according to the invention. A radio-frequency part **24** is in accordance with the prior art and comprises signal reception and amplification elements. An A/D converter **25** converts an analog signal to a digital one and feeds it to the decoding and reproduction part **26** in order to decode the digital information and reproduce the program conveyed by it. An pulse response estimation block **27** calculates a Fourier transform for a sequence of samples produced by the A/D converter, multiplies it by a complex conjugate of the Fourier transform of the known signal read from the memory **28** and inverse-transforms the result, thus producing the pulse response curve. A location block **29** finds the maximum and the beginning and the end for the pulse response. A symbol synchronization block **30** estimates the length of the pulse response and informs the A/D converter **25** about the optimum location of the guard interval with respect to the pulse response and corrects the time indexes in the memory indicating the beginning and end of the pulse response so that they correspond to the new location of the guard interval. A sampling frequency adjustment block **31** chooses certain differences indicating the shift of the pulse response as the basis for a sampling frequency correction and sends a correction instruction to the A/D converter **25**. All procedures described above are preferably implemented by programming them as instructions to be carried out by a microprocessor in a manner known to a person skilled in the art.

The invention provides a practical and computationally relatively light method for adjusting the symbol synchronization and sampling frequency in an OFDM receiver. The method according to the invention can be implemented using a receiver apparatus based on common components, so it is suitable for mass production at a cost level required for consumer electronics.

What is claimed is:

1. A method for synchronizing a receiver to an orthogonal frequency division multiplex (OFDM) signal transmitted on a radio channel, said OFDM signal comprising successive symbols separated by guard intervals, comprising the steps of:

determining (**12**) a pulse response of the radio channel on which the receiver is receiving the transmitted OFDM signal as a received OFDM signal;

setting (**17**; **18**) a guard interval time to cause the received OFDM signal to coincide with the most significant components of a pulse response corresponding to one symbol;

examining (**14**) the guard interval time for determining a length of the pulse response with respect to the length of the guard interval, and

comparing pulse response beginning and end points obtained from two successive pulse response estimation rounds to each other and choosing an earlier of beginning points obtained from the two successive estimation rounds as a pulse response beginning point, and choosing a later of the end points obtained from the two successive estimation rounds as the pulse response end point.

2. The method of claim 1, wherein said step of setting comprises as mutually exclusive alternatives, steps wherein

if the pulse response is shorter than the guard interval setting (17), the guard interval time to begin at a beginning of the pulse response from a moment where a pulse response value for a first time exceeds a certain first threshold value, and

if the pulse response is longer the guard interval setting (18), the guard interval time to begin with respect to the pulse response so that a sum of absolute values of the pulse response components left outside the guard interval is as small as possible.

3. The method of claim 1, further comprising the step of measuring (20) a temporal shift of the pulse response between pulse measurements and correcting a receiver sampling frequency on a basis of the measured temporal shift.

4. The method of claim 3, further comprising a step of discriminating (21) and eliminating pulse response shifts that have the highest absolute values for correcting the sampling frequency.

5. The method of claim 4, further comprising the steps of storing and arranging a certain first quantity of successive measured pulse response shifts in order of magnitude, and selecting a certain second quantity of shifts from a middle of a resulting sequence for correcting the sampling frequency.

6. The method of claim 7, wherein said first quantity is nine and said second quantity is three.

7. A receiver for receiving a digital orthogonal frequency division multiplex (OFDM) modulated transmission comprising symbols separated by guard intervals on a radio channel having a certain variable pulse response, comprising:

means (27) for determining an instantaneous pulse response of the radio channel;

a synchronizable analog-to-digital (A/D) conversion circuit (25), operation of which includes cyclically repeated guard interval time and information time;

means (27) for examining the guard interval time for determining a length of the pulse response with respect to the length of the guard interval;

means (28,29) for comparing pulse response beginning and end points obtained from two successive pulse response estimation rounds to each other to choose an earlier of beginning points obtained from said two successive estimation rounds as a pulse response beginning point, and to choose a later of the end points obtained from said two successive estimation rounds as the pulse response end point, and

means (30) for driving said A/D conversion circuit to a state in which said guard interval time covers a period in the calculated pulse response which contains the most significant components of the pulse response.

8. The receiver of claim 7, further comprising means (31) for determining a temporal shift of the pulse response occurring between different pulse response determinations.

9. The receiver of claim 8, further comprising means (31) for changing a sampling frequency used by said A/D conversion circuit on the basis of a calculated temporal shift of the pulse response.

10. A method for synchronizing a receiver to an orthogonal frequency division multiplex (OFDM) signal transmitted on a radio channel, said OFDM signal comprising successive symbols separated by guard intervals, comprising the steps of:

determining (12) a pulse response of the radio channel on which the receiver is receiving the transmitted OFDM signal as a received OFDM signal;

setting (17; 18) a guard interval time to cause the received OFDM signal to coincide with the most significant components of a pulse response corresponding to one symbol;

measuring (20) a temporal shift of the pulse response between pulse measurements and correcting a receiver sampling frequency on a basis of the measured temporal shift

discriminating (21) and eliminating pulse response shifts that have the highest absolute values for correcting the sampling frequency;

storing and arranging a certain first quantity of successive measured pulse response shifts in order of magnitude, and

selecting a certain second quantity of shifts from a middle of a resulting sequence for correcting the sampling frequency.

11. The method of claim 10, wherein said first quantity is nine and said second quantity is three.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE

CERTIFICATE OF CORRECTION

PATENT NO : 6,125,124

DATED : September 26, 2000

INVENTOR(S) : Junell et al

It is certified that error appears in the above-identified patent and that said Letters Patent are hereby corrected as shown below:

At col. 9, line 26 (claim 6, line 1), please cancel "7" and substitute --5-- therefor.

At col. 10, line 31 (claim 10, line 16), after "shift", please insert --;--.

Signed and Sealed this

Eighth Day of May, 2001

Attest:

Nicholas P. Godici

NICHOLAS P. GODICI

Attesting Officer

Acting Director of the United States Patent and Trademark Office

US006282167B1

(12) **United States Patent**
Michon et al.

(10) **Patent No.: US 6,282,167 B1**
(45) **Date of Patent: *Aug. 28, 2001**

(54) **OFDM SIGNAL ORGANIZED SO AS TO SIMPLIFY RECEPTION**

(75) Inventors: **Vincent Michon**; **Bernard Le Floch**; **Bertrand Sueur**, all of Rennes; **Marc Riviere**, Liffre, all of (FR)

(73) Assignee: **France Telecom**, Paris (FR)

(*) Notice: This patent issued on a continued prosecution application filed under 37 CFR 1.53(d), and is subject to the twenty year patent term provisions of 35 U.S.C. 154(a)(2).

Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

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(22) PCT Filed: **Jun. 13, 1995**

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§ 371 Date: **Mar. 19, 1997**

§ 102(e) Date: **Mar. 19, 1997**

(87) PCT Pub. No.: **WO95/34967**

PCT Pub. Date: **Dec. 21, 1995**

(30) **Foreign Application Priority Data**

Jun. 16, 1994 (FR) 94 07612

(51) **Int. Cl.**[7] **H04J 11/00**

(52) **U.S. Cl.** **370/203**; 370/482

(58) **Field of Search** 370/203, 206, 370/208, 210, 344, 489, 482, 484, 486, 209; 375/130, 242

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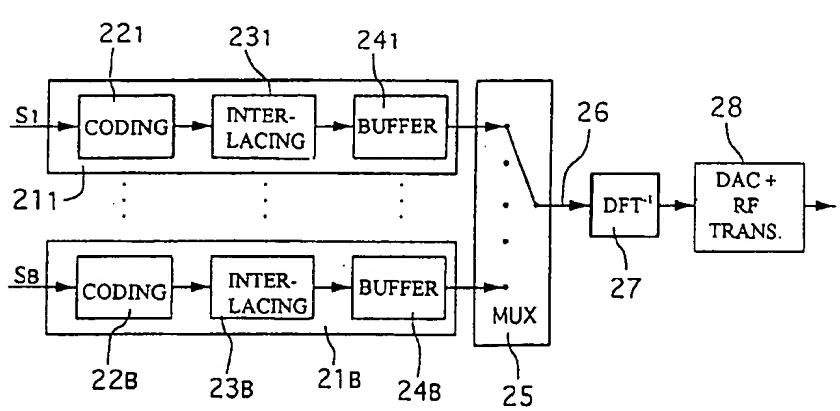
Primary Examiner—Joe H. Cheng
Assistant Examiner—Kim T. Nguyen
(74) *Attorney, Agent, or Firm*—Kinney & Lange, P.A.

(57) **ABSTRACT**

The invention relates to a signal intended to be transmitted towards a plurality of receivers, of the type comprising at least two source signals and consisting of a plurality of independently modulated substantially orthogonal carrier frequencies distributed over a predetermined frequency band. According to the invention, said frequency band **(12)** is divided into at least two frequency sub-bands (13_1–13_4) each comprising a set of said substantially orthogonal carrier frequencies **(11)**, and to each of said sub-bands is allocated one of said source signals, so that a receiver may retrieve from the transmitted signal, by filtration, at least one of said sub-bands and perform demodulation processing only on the carrier frequencies contained in the retrieved sub-bands. The invention also relates to a transmission process as well as to a receiver of such signal.

14 Claims, 2 Drawing Sheets



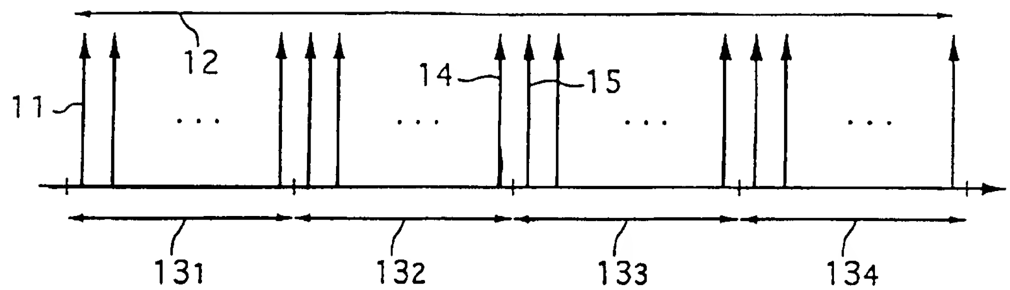


Fig. 1

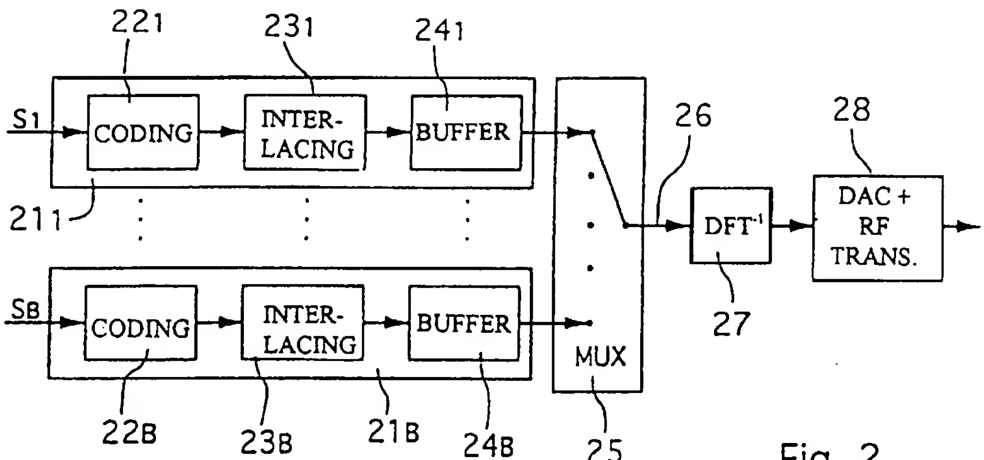


Fig. 2

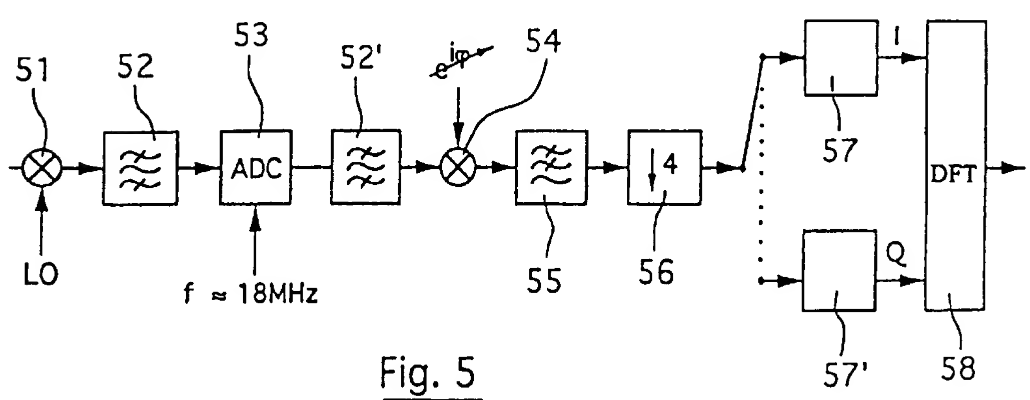


Fig. 5

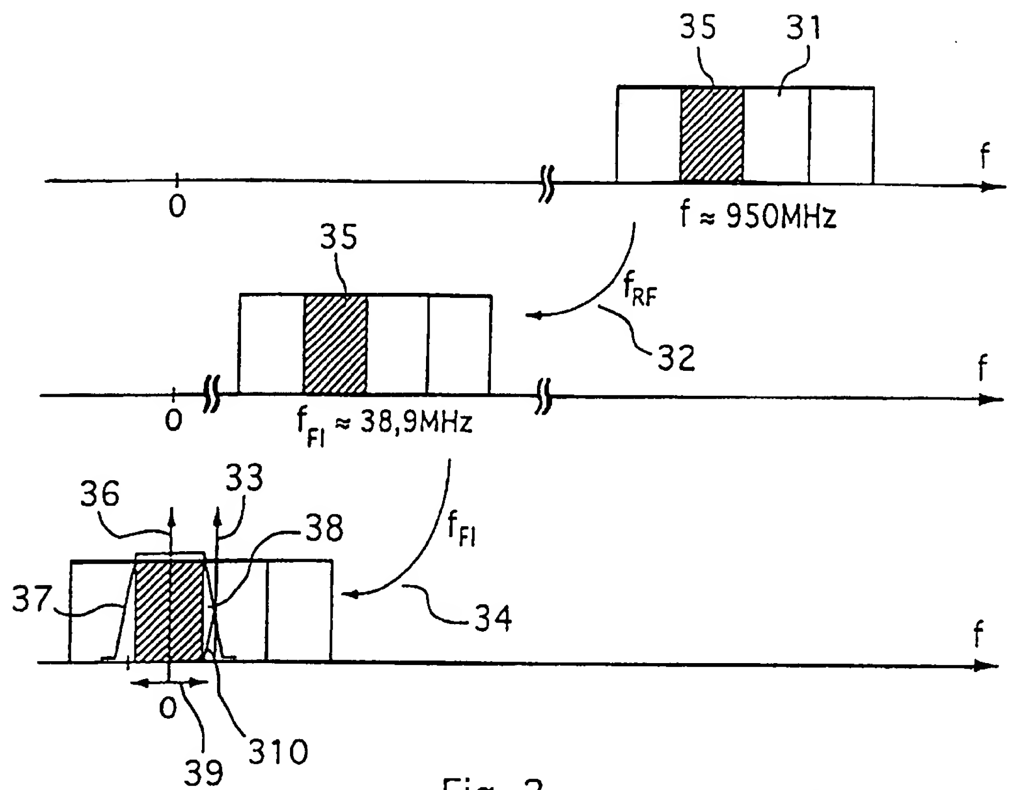

35
31
f
0
f ≈ 950MHz
35
f_{RF}
32
f
0
f_{FI} ≈ 38,9MHz
36
33
38
f_{FI}
37
34
f
0
39
310

Fig. 3

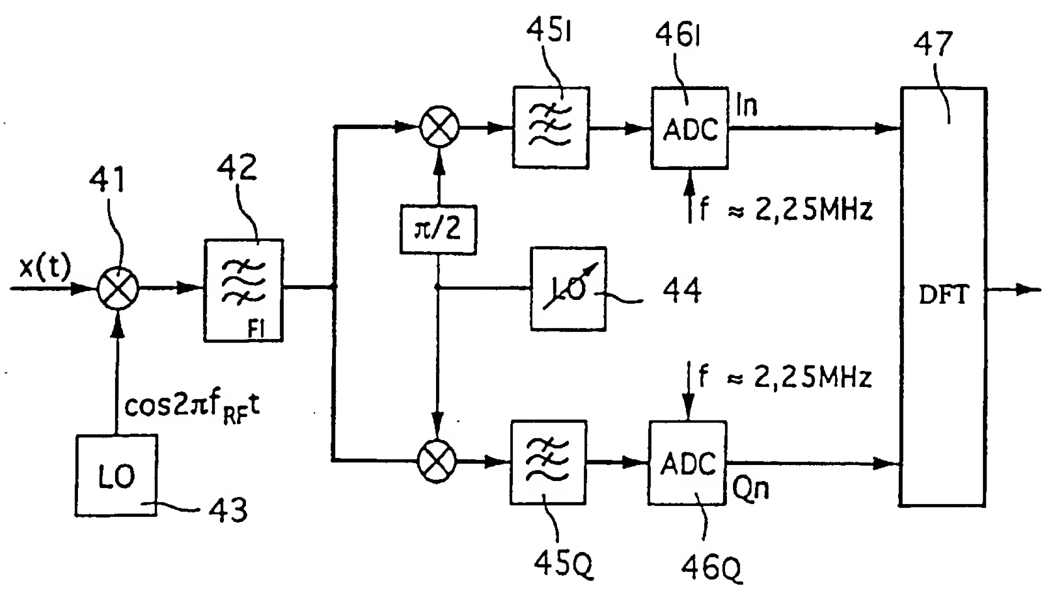

45I
46I
47
ADC
In
41
42
π/2
f ≈ 2,25MHz
x(t)
FI
LO
44
DFT
f ≈ 2,25MHz
$\cos 2\pi f_{RF} t$
ADC
Qn
LO
43
45Q
46Q

Fig. 4

OFDM SIGNAL ORGANIZED SO AS TO SIMPLIFY RECEPTION

BACKGROUND OF THE INVENTION

The field of the invention is signal transmission using simultaneously several orthogonal (or quasi-orthogonal) carrier frequencies, each coded by distinct data elements.

These signals are generally called OFDM (Orthogonal Frequency Division Multiplex) signals. This type of OFDM signal is used for example in the digital broadcasting system described particularly in French patent FR-86 096322 filed on Jul. 2, 1986, and in the document entitled "Principes de modulation et de codage canal en radio diffusion numeric vers les mobiles" (Principles of channel modulation and coding in digital radio broadcasting to mobiles) (by M. Alard and R. Lassalle; U.E.R. review No. 224, August 1987, pp. 168–190) and known under the name of the COFDM (Coded Orthogonal Frequency Division Multiplex) system.

This COFDM system was developed largely as part of the European DAB (Digital Audio Broadcasting) project. It is digital. More generally, it enables the transmission of any type of digital or analog signal (sampled but not necessarily quantified).

Special demodulators must be used to demodulate these digital signals with frequency multiplexing. For example, this type of demodulator is described in the above mentioned patent document FR-86 09622.

It is known that one essential element of a multicarrier signal receiver is the demodulation circuit which extracts raw information carried by each carrier taken separately, from the received signal (the multiplex of orthogonal carriers).

Conventionally, this circuit carries out mathematical transform the signal, and for example a Discrete Fourier Transform (DFT). Many other transforms may be used. However, this type of circuit will be referred to as a DFT circuit in the following, for non-restrictive simplification purposes.

The complexity of this type of circuit is proportional firstly to the number of frequencies transmitted simultaneously (frequency dimension), and secondly to the duration T_s of transmitted symbols (time dimension). This DFT circuit is a complex and therefore expensive element. Therefore, it is essential that this circuit should be simplified, particularly for low cost receivers.

According to known techniques, the time dimension is limited by reducing the symbol time T_s and/or the guard interval Δ inserted between two consecutive symbols. This limits the number of data processed by the DFT, obviously to the detriment of the received signal quality. As the symbol time increases, the channel selectivity effect is lower, and for a given guard interval sufficient to limit the Inter Symbol Interference to a previously chosen value, the transmitted throughput increases with the length of the symbol time.

In other words, the choice of the DFT size is always a compromise between the received signal quality and the cost price of this DFT.

It is impossible to vary the frequency dimension using known techniques. The DFT must systematically take account of all N carriers forming the transmitted multiplex, even if the information searched by the receiver is distributed only on some of the carrier frequencies.

Conventionally, a transmitted OFDM signal can carry several independent signals. For example in the case of television signals, four distinct signals could be transmitted at 6 Mbit/s on one OFDM signal occupying an 8 MHz band (with a spectral efficiency of 4 bits/s/Hz). Although it would be desirable to recover a single source signal, it is necessary to take account of all carriers at the input to the DFT, which induces complex and partially unnecessary calculations.

Furthermore, in general it is better to use the maximum number of carrier frequencies, particularly to increase the duration of symbols transmitted as described previously.

Once again, we need to find a compromise between the number of carriers and the complexity of the demodulation circuit.

BRIEF SUMMARY OF THE INVENTION

The invention has notably the objective of overcoming these drawbacks in state of the art techniques.

More precisely, a purpose of the invention is to provide a technique that reduces processing to be done in receivers without any loss of signal quality, in other words particularly without it being necessary to reduce the symbol time.

Another purpose of the invention is to provide this type of technique without inducing excessive constraints on the transmission, and particularly in which it is not necessary to widen the frequency band used.

Another purpose of the invention is to provide this type of technique in order to define several receiver quality levels.

These objectives, and others that will become apparent later, are achieved according to the invention by a signal to be transmitted to several receivers, of the type comprising at least two source signals and composed of several substantially orthogonal carrier frequencies modulated independently and distributed on a determined frequency band; said frequency band being organized down into at least two frequency subbands each comprising a set of said substantially orthogonal carrier frequencies, and in which one of said source signals is assigned to each of said subbands, so that a receiver can extract at least one of said subbands from the transmitted signal by filtering, and can carry out demodulation processing solely on carrier frequencies contained in the extracted subbands.

Subbands refer to carrier subassemblies that are not necessarily separate (one carrier may belong to several subbands), or contiguous (two carriers in one subband may be separated by one or several carriers that do not belong to this subband). Some source signals could also be "empty", in other words associated with non-transmitted carriers.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 shows a signal according to the present invention corresponding to four independent television signals at 6 Mbit/sec each.

FIG. 2 shows a block diagram of a transmitter building and transmitting the signal illustrated in FIG. 1.

FIG. 3 illustrates the principle of the transposition of the signal into the base band according to the present invention.

FIG. 4 shows an analog demodulator according to the present invention.

FIG. 5 shows a digital demodulator according to the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Thus the invention concerns a new multiplexed signal structure with orthogonal carriers in order to limit processing carried out in receivers.

More precisely, the invention decorrelates the number of carriers Q processed in the receiver from the number P ($Q<P$) of carriers transmitted. Thus this gives the advantages of a "large" inverse DFT (P points) during transmission, and a "small" DFT (Q points) on reception.

Note that this signal structure is unrelated to the conventional technique consisting of transmitting each source signal independently, in distinct and separate frequency bands.

In this case, different signals would be transmitted by apparently different transmitters. This can induce very large power differences, and therefore disturbances, between two signals. On the contrary, according to the invention a single signal is tuned and transmitted.

Furthermore, according to known techniques, it is necessary to create a very wide frequency band (of the order of 1 MHz) between two signals in order to limit interference (orthogonality by separate frequency supports). On the contrary according to the invention, subbands may be adjacent due to the fact that the transmitted signal is tuned as a whole, and carriers are orthogonal, even from subband to subband.

This signal structure is not at all obvious, looking at known OFDM signals. Firstly, it seems to contradict the idea that it is preferable to distribute data over the widest possible frequency band in order to benefit from the best frequency diversity. Furthermore, it cannot be used directly in OFDM receivers. As will be described later, the transposition of the received signal has to be controlled to center the required subband(s).

It is beneficial if said subbands are adjacent.

Preferably, said subbands will have identical bandwidths. This can simplify processing in receivers, and more easily maintain a good frequency diversity at the transmission, provided that a suitable decoding, preferably with maximum probability, is implemented independently on each subband.

Optionally, assignment of said source signals to said subbands may be variable in time, in order to improve the frequency diversity.

For example, said assignment may be modified at each frame of said signal. A frame is a set of one or several symbols.

In particular, it is possible that the transmitter will change the assignment for each new frame. This can counter selective fading affecting each of the subbands by maximum frequency diversity enabled by the band occupied by all these subbands.

According to one beneficial embodiment of the invention, at least a first of said source signals contains basic information for a program, and at least a second of said source signals corresponds to information complementary to said basic information, such that at least two receiver quality levels are defined:

- a first quality level corresponding to receivers capable of processing only the subband corresponding to said source signal;
- a second quality level corresponding to receivers capable of processing subbands corresponding to the first and second source signals.

A program means one or several source signals (or components), in which the subbands of which are preferably adjacent. Thus a program may have several components, and conversely a component may belong to several programs.

In particular, this means that several receiver quality levels can be defined. For example in the case of television signals, it will be possible to allow for a first subband containing information used to restore a medium quality image and a second subband including either complementary information enabling cooperation with information in the first subband, to restore a high definition image, or information used to rebuild this second image.

Thus, at least three types of receiver may be defined:

- an entry quality receiver, which only processes the first subband;
- a medium quality receiver which processes either the first or the second subbands (preferably close in the frequencies space);
- a high quality receiver processing the entire transmitted signal, and allowing either simultaneous display of several programs, for example using the picture in picture technique, or reproduction of the original high definition image.

The invention also concerns a process for transmitting a signal like that described above including the following steps:

- assignment of a determined frequency band to said signal, where several orthogonal carrier frequencies are defined in the frequency band;
- breakdown of said frequency band into at least two frequency subbands, each comprising a set of said approximately orthogonal carrier frequencies;
- reception of at least two independent source signals to be transmitted;
- assignment of one of said frequency subbands to each of said source signals;
- grouping of said subbands, so as to form said signals to be transmitted; and
- transmission of said signal to be transmitted.

It is beneficial if said subbands are adjacent.

Preferably, said subband grouping step is preceded by an independent coding and frequency and time interlacing step for each of said source signals, so as to obtain a set of coded signals each of which modulates one of said carrier frequencies of the subband assigned to said source signal.

The invention also concerns receivers of this type of signal. Advantageously, these receivers comprise:

- means for selecting a given program, corresponding to one of said subbands; and
- mathematical transformation means acting on carrier frequencies contained in the selected subband(s).

According to one essential characteristic of the invention, means for selecting a given program are capable of transposing the received signal, which is not a fixed operation (unlike classical techniques). On the contrary, this transposition is offset by an amount that depends on the required subbands, before the DFT is applied. Obviously, this DFT only applies to extracted subbands, which correspondingly reduces the processing to be done.

According to a first analog type of embodiment, said selection means include analogue transposition means comprising a first RF transposition oscillator and a second IF transposition oscillator, and means for controlling the oscillation frequency of said first and/or said second oscillator as a function of the selected subband(s), so that they will be centered at a predetermined frequency.

According to a second digital embodiment, said selection means include first analog digital transposition means, and second digital transposition means that are variable as a function of the selected subband(s) and subsampling means.

Preferably, said mathematical transformation means act on a number of carrier frequencies that is slightly greater than the number of carrier frequencies contained in the

extracted subband(s), in order to compensate for the imperfection due to extraction filtering of said subbands. In particular an imperfection means folding over of residual spectra.

Other characteristics and advantages of the invention will become clear in reading the following description of a preferred embodiment of the invention which is given as a simple and non-restrictive example for illustration purposes only, and the joint drawings, in which:

As described above, the invention concerns a signal formed from several orthogonal frequencies. The embodiment described below corresponds to broadcasting of four television signals according to the COFDM technique mentioned above.

Obviously, this is just an example; the number and size of sub-bands, the type of source signals and the transmission technique used may vary.

Therefore the example in FIG. 1 considers an OFDM signal, conventionally containing 8 192 carriers 11(of which 7 000 are useful), distributed over a frequency range 12 of 9 MHz. For coding at a rate of 4 bit/s/Hz, it is then possible to transmit about 24 Mbit/s.

According to the invention, the frequency band **12** is broken into 4 subbands, or blocks $\mathbf{13}_1$ to $\mathbf{13}_4$, each of which can carry 6 Mbit/s. These 6 Mbit/s may each correspond to a standard television program. Obviously, the number and size of these blocks are simply given for guidance.

Each block $\mathbf{13}_i$ carries data corresponding to an independent or standalone signal. In other words, there is no need to recover carrier frequencies other than those in the block considered, in order to rebuild this signal. However no separation is necessary, for example, between carriers **14** and **15**.

However, note that a standalone signal does not necessarily correspond directly to a program (for example television). Complementary signals may also be used to improve the quality of the basic signal, or more generally any signal forming part of a set.

In the case of a COFDM signal, the signal transmitted in RF is formed of a time sequence of symbols of duration $Ts=ts+\Delta$ where ts is the duration of the useful symbol (ts=NT), on which the demodulation will be applied and where Δ represents the duration of the guard interval.

Each symbol is then written:

$$x(t)=\sum_{i=0}^{B-1}x_i(t)=\sum_{i=0}^{B-1}\mathrm{Re}\sum_{k\in K_i}C_k e^{2i\pi f_k t}$$

where $t\in[-\Delta, ts]$

and $f_k=f_0+k/ts$

where C_k=element of the modulation alphabet (finite or not)

f_0=arbitrary frequency

N=total number of tuned carriers. For practical reasons, N is often integer power of 2 (for example N=8192) greater than the number of carriers actually modulated. The additional (N−P) carriers (called "untransmitted") are then modulated by 0(C_k=0 for k∈ξξ being the set of useful carriers in the signal.

Or

$$x(t)=\sum_{k=0}^{N-1}C_k e^{2i\pi f_k t}$$

where: B is the number of blocks

K_i is the set of integers $[k_i, k_{i+1}]$

where $k_0=0$ and $k_{i+1}=k_i+Q_i$(Q_i being arbitrary, number of carriers in block Bi).

For example, the signal may be formed from 8 192 carriers organized in 6 blocks, of which only 4 are useful:

B_0 formed of carriers k_0=0 to 595, carriers not transmitted;

B_1 formed of carriers k_1=596 to 2345, with carriers assigned to a first program;

B_2 formed of carriers k_2=2346 to 4095, with carriers assigned to a second program;

B_3 formed of carriers k_3=4096 to 5845, with carriers assigned to a third program;

B_4 formed of carriers k_4=5846 to 7595, with carriers assigned to a fourth program;

B_5 formed of carriers k_5=7596 to 8191, not transmitted;

FIG. 2 presents the general block diagram of a transmitter capable of generating and transmitting a signal according to the invention.

It comprises firstly four parallel independent lines $\mathbf{21}_1$ to $\mathbf{21}_B$ corresponding to B source signals, and B blocks in the signal for this particular example.

Each source signal S_i(where I varies from 0 to B−1) is subjected to signal coding $\mathbf{22}_i$, in the case of a digital signal. For example, the coding described in the Alard and Lassalle document mentioned above may be used. An analog signal will simply be sampled.

The data are then advantageously interlaced in frequency and/or in time ($\mathbf{23}_i$). Obviously, this interlacing is done in a stable manner, in other words such that data in signal S_i remain in the block assigned to S_i.

Data are then stored in buffer memories $\mathbf{24}_i$ which make the complete signal formed from a series of symbols C_k **26** by frequential multiplexing **25** (in other words successive reading of each block in buffer $\mathbf{24}_i$).

The order in which buffers are read may be different in each frame (according to a known receiver sequence). It is thus possible to maintain the frequency diversity qualities of a conventional COFDM signal.

The C_k symbols are then processed conventionally, by inverse Fourier transformation (DFT^{-1})**27**, then by digital/analog conversion, transposition in RF and transmission **28**. As mentioned previously, a single signal is transmitted. Therefore, there is no significant power difference between two blocks.

Due to the signal structure, it is possible to use a transformation in reception acting on a smaller number of points than the inverse transformation used in transmission.

In doing this, the transposition in low frequency done by the receivers is significantly different from that done conventionally, as shown in FIG. **3** in the case of an analogue transposition.

Conventionally, signal **31** is transmitted in radio frequency (RF), for example at 950 MHz. Therefore, it is subjected to a first multiplication **32** by a frequency F_{RF} which brings it to an intermediate frequency (IF) for example at 38.9 MHz. A second multiplication **34** by a frequency f_{if} changes the signal to low frequency.

Conventionally and for a given RF channel, the frequencies F_{RF} and f_{IF} are fixed. However, according to an analogue embodiment of the invention, one of them must be variable.

It is required to process only one signal block, for example block **35**. Therefore, the transposition frequencies are adjusted so that this block **35** is centered on the zero frequency **36** after transposition (although usually the complete signal is centered). For example, if f_{iF} is adjusted, this frequency could be f_{IF}=38.9±f_i where f_i depends on which block is selected.

After transposition, the selected block is filtered (**37**). In order to select all useful data, a filter pattern has to be used which also includes useless elements (attenuated band due to filtering **38** which is necessarily not rectangular which would be ideal). Consequently, a slightly wider transformation **39** will also be applied.

More precisely, a sampling frequency **33** will be used (for example 2.25 MHz) slightly greater than the block width (for example 1.92 MHz), in order to force oversampling. Consequently, transformation **39** is defined so as to encompass half of the attenuated band **38**. Thus, spectrum **310** is folded back and remains in the attenuated band zone, and therefore does not pollute the useful signal. For example, if block **35** contains 1750 points, the transformation **89** will apply to 2048 points.

FIG. **4** illustrates the case of an analogue demodulator making use of this technique.

The received signal x (t) is conventionally transposed from RF to IF by the demodulator **41** which multiplies x(t) by cos $2\pi f_{RF}t$ and is then filtered by a FOS filter **42** centered on the IF frequency, for example of the order of 35 MHz. The frequency f_{RF} is output by an RF oscillator **43** with an adjustable tuner frequency used to select the RF channel.

The second transposition is then done, which is variable as a function of the selected block. It is done by a second variable oscillator **44** adjustable within the $f_0, \ldots f_{B-1}$ band in order to bring the required block into the base band.

The demodulation then takes place conventionally, to obtain two channels I_n and Q_n after low pass filtering 45_I and 45_Q and sampling 46_I and 46_Q, sampled at frequency f_s (for example 2.25 MHz) which are input to a DFT circuit **47** applied only to the number of points (or slightly more) making up the block (for example 2048).

The RF and IF frequencies are not affected in the case of a digital transposition as shown in FIG. **5**. However, sampling is done at a much higher frequency (for example eight times higher) than in analogue.

Thus, the signal is conventionally transposed (**51**), filtered (**52**), digitized (**53**) at a frequency f_s of the order of 18 MHz, and is then filtered again (**52'**).

A block is then selected by multiplying **54** each sample by $e^{i\phi}(t)$ where $\phi(t)$ depends on the required block. The signal obtained is filtered by a low pass filter **55** and is then subsampled **56** by order **4** in order to recover the required signal which is input to a DFT circuit **58** after interpolation filtering **57, 57'**.

What is claimed is:

1. A method for transmitting and receiving at least two independent source signals, comprising the steps of:
 - obtaining said at least two independent source signals in the form of independent series of coded bits;
 - assignment of a determined frequency band to an OFDM signal to be transmitted, several approximately orthogonal carrier frequencies being defined in said frequency band;
 - breakdown of said frequency band into at least two frequency subbands, each of said subbands comprising a set of said approximately orthogonal carrier frequencies;
 - assignment of each of said frequency subbands to one of said independent source signals;
 - selectively modulating the carrier frequencies of each frequency subband with the coded bits of the corresponding source signal;
 - grouping said modulated frequency subbands to form a modulated OFDM signal;
 - tuning and transmitting the modulated OFDM signal as a whole;
 - receiving the modulated OFDM signal in a receiver;
 - extracting from the modulated OFDM signal at least one, but not all the frequency subbands, by filtering; and
 - performing demodulation processing solely on the frequency carriers contained in the extracted subbands of the modulated OFDM signal.

2. Method according to claim **1**, characterized in that said subbands are adjacent.

3. Method according to claim **1**, characterized in that said subband grouping step is preceded by an independent coding step and frequency and time interlacing of each of said source signals, so as to obtain a set of coded signals designed to modulate each of said carrier frequencies of the subband assigned to said source signal.

4. Method according to claim **1**, characterized in that said source signals are assigned to said subbands in a manner that varies with time, in order to maximize the frequency diversity.

5. Method according to claim **4**, characterized in that said assignment is modified on each transmission of a frame of said signal.

6. Method according to claim **1**, wherein the modulated OFDM signal is a single signal tuned as a whole by a sole modulator modulating simultaneously the substantially orthogonal frequency carriers, the orthogonal frequency carriers being orthogonal in each subband and from subband to subband.

7. Method according to **6**, characterized in that said subbands are adjacent.

8. Method according to claim **6**, characterized in that said subbands have identical bandwidths.

9. Method according to claim **1**, characterized in that at least a first of said source signals corresponds to basic information for a program and at least a second of said source signals corresponds to information complementary to said basic information, in order to define at least two receiver quality levels:
 - a first quality level applicable to receivers capable of processing only the subband corresponding to said first source signals; and
 - a second quality level corresponding to receivers capable of processing subbands corresponding to the first and second source signals.

10. Method according to claim **1**, characterized in that performing demodulation processing further comprises:
 - selecting a given program corresponding to at least one of the frequency subbands using a selection means; and
 - acting on the carrier frequencies contained in the selected subband(s) using a mathematical transformation means.

11. Method according to claim **10**, characterized in that said selection means include analog transposition means including a first RF transposition oscillator and a second IF transposition oscillator, and means of controlling an oscillation frequency of said first RF transposition oscillator and/or said second IF transposition oscillator as a function of the selected subbands, so that the selected subbands are centered on a predetermined frequency.

12. Method according to claim **10**, characterized in that said selection means comprises:

first analog transposition means and second digital transposition means that are variable as a function of the selected subband(s); and

subsampling means.

13. Method according to claim **10**, characterized in that said mathematical transformation means act on a number of carrier frequencies slightly exceeding the number of carrier frequencies contained in the extracted subband(s), so as to compensate for imperfection due to extraction filtering of said subbands.

14. A method for transmitting and receiving an OFDM signal, the method comprising:

obtaining at least two independent source signals, each source signal being in the form of an independent series of coded bits;

assigning a determined frequency band on which the OFDM signal will be transmitted;

defining approximately orthogonal carrier frequencies in the determined frequency band;

breaking the determined frequency band down into at least two frequency subbands, each of said subbands

comprising a set of said approximately orthogonal carrier frequencies;

assigning each independent source signal to one of said frequency subbands;

transmitting a modulated OFDM signal by selectively modulating the carrier frequencies of each frequency subband with the coded bits of the correspondingly assigned source signal and grouping said modulated frequency subbands to form said modulated OFDM signal, said modulated OFDM signal being tuned and transmitted as a whole, so that said frequency carriers are orthogonal in each of said subbands and from subband to subband;

receiving the modulated OFDM signal;

extracting at least one but less than all of the frequency subbands from the received OFDM signal by filtering; and

performing demodulation processing solely on the frequency carriers contained in the extracted subbands of the received modulated OFDM signal.

* * * * *

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Kim et al.

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(45) **Date of Patent: Jan. 9, 2001**

(54) **FRAME SYNCHRONIZATION METHOD AND APPARATUS FOR USE IN DIGITAL COMMUNICATION SYSTEM UTILIZING OFDM METHOD**

(75) Inventors: **Young-Sang Kim**, Soeul (KR); **Francois Langinieux**, Metz (FR)

(73) Assignee: **Daewoo Electronics Co., Ltd.**, Seoul (KR)

(*) Notice: Under 35 U.S.C. 154(b), the term of this patent shall be extended for 0 days.

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(22) Filed: **Dec. 1, 1997**

(30) **Foreign Application Priority Data**

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(51) **Int. Cl.**^{7} **H04J 3/06**
(52) **U.S. Cl.** **370/516**; 375/354
(58) **Field of Search** 370/509, 510, 370/511, 512, 516, 491, 492; 375/232, 235, 354, 366, 364; 348/425.4, 500, 501, 507, 513

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Primary Examiner—Chau Nguyen
Assistant Examiner—Chiho Andrew Lee
(74) *Attorney, Agent, or Firm*—Pillsbury Madison & Sutro LLP

(57) **ABSTRACT**

A frame synchronization method and apparatus for use in a digital communication system utilizing OFDM method are disclosed. The frame synchronization apparatus comprises a phase calculator for calculating phase values of TPS pilots within a symbol according to in-phase and quadrature-phase channel signals received from a transmitting side; a D-BPSK demodulator for performing D-BPSK demodulation for the phase values of TPS pilots supplied from the phase calculator and outputting the TPS pilots within the demodulated symbol; a control signal generator for comparing the demodulated TPS pilots with each other and outputting a control signal according to the compared result; and frame synchronization unit for confirming a sync word position according to the control signal supplied from the control signal generator and outputting a frame sync signal. Hence, frame synchronization can be achieved by using the synchronization word inverted at each frame in one TPS block without the need to increase the size of hardware.

12 Claims, 4 Drawing Sheets

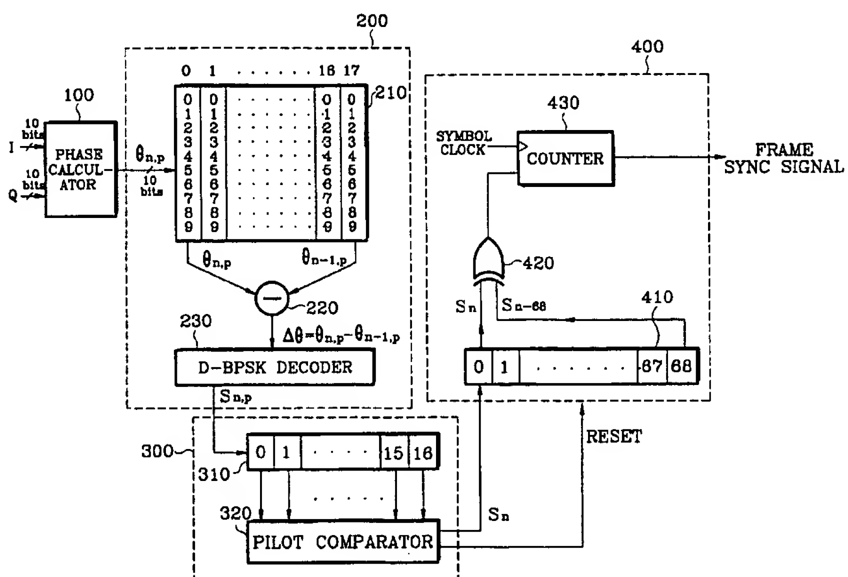


FIG. 1A(PRIOR ART)

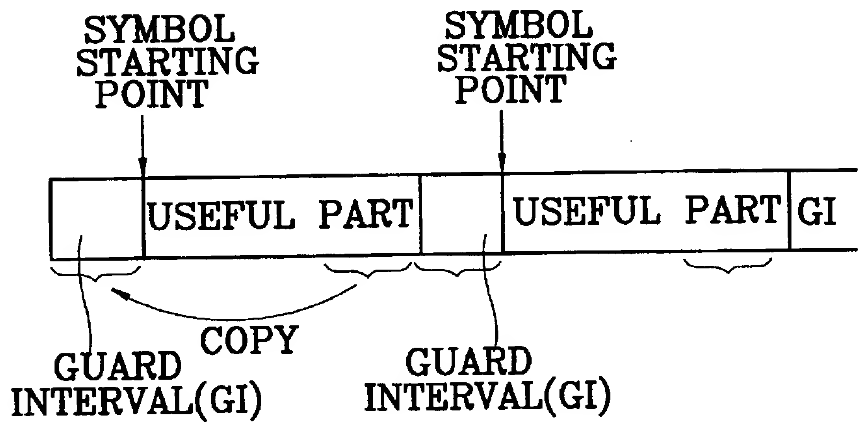


FIG. 1B(PRIOR ART)

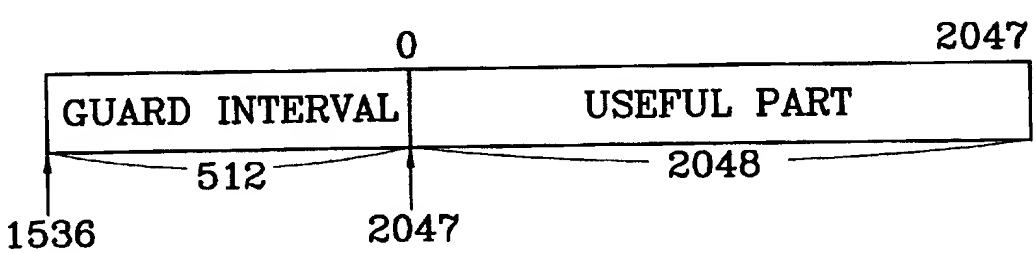


FIG.2

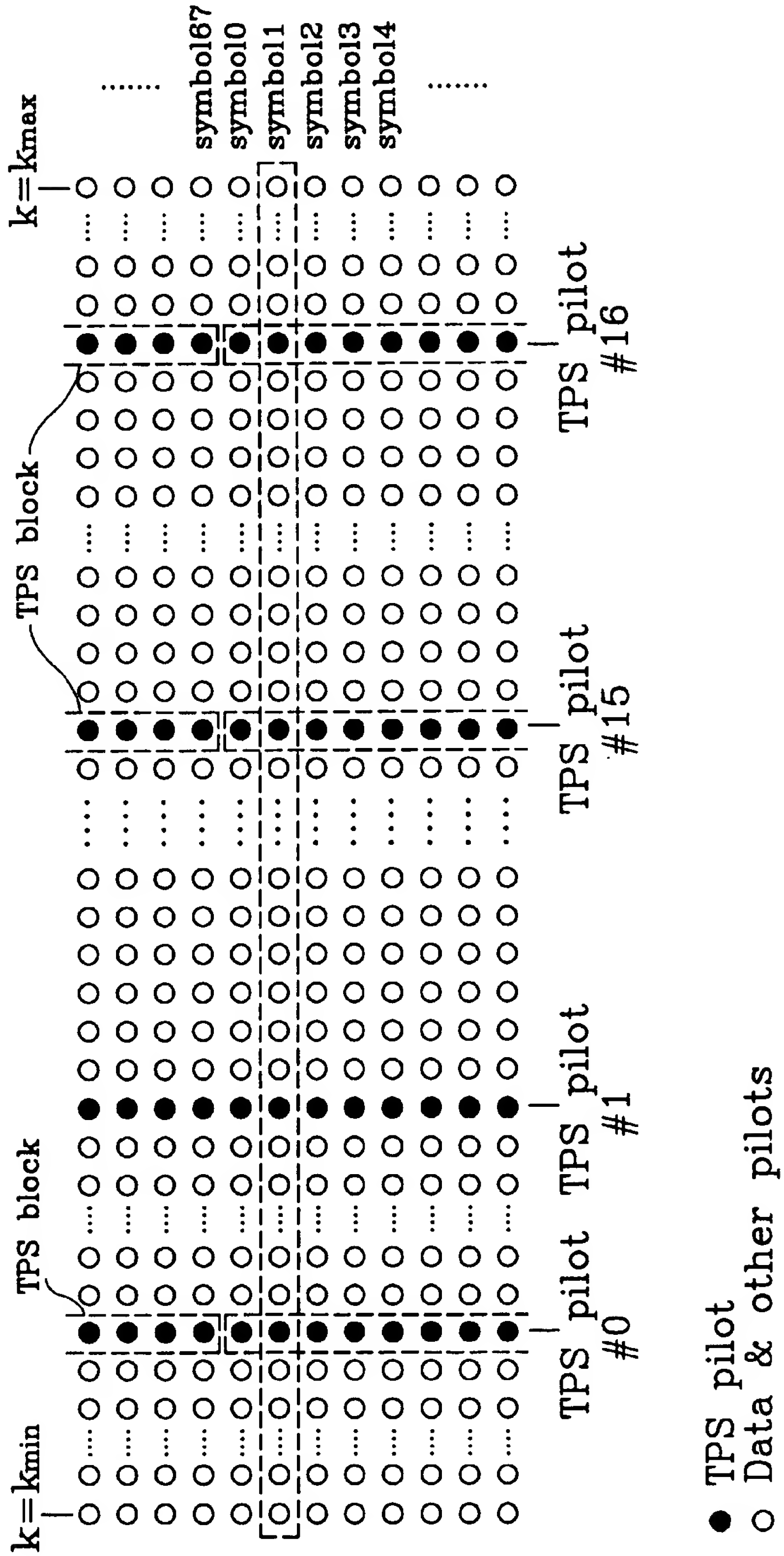

k=kmax
k=kmin
symbol67
symbol0
symbol1
symbol2
symbol3
symbol4
TPS block
TPS block
TPS pilot #0
TPS pilot #1
TPS pilot #15
TPS pilot #16
TPS pilot
Data & other pilots

FIG.3

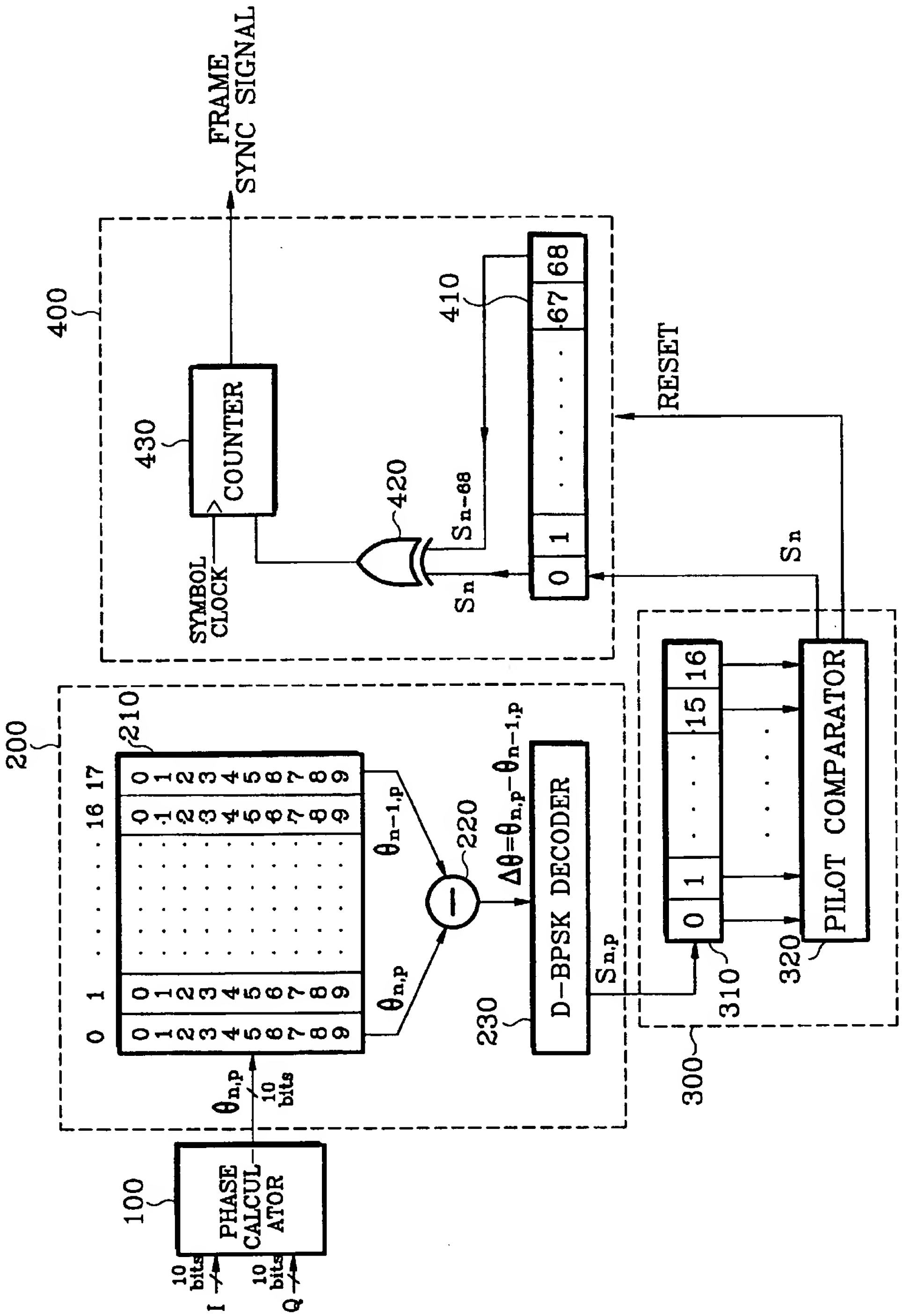
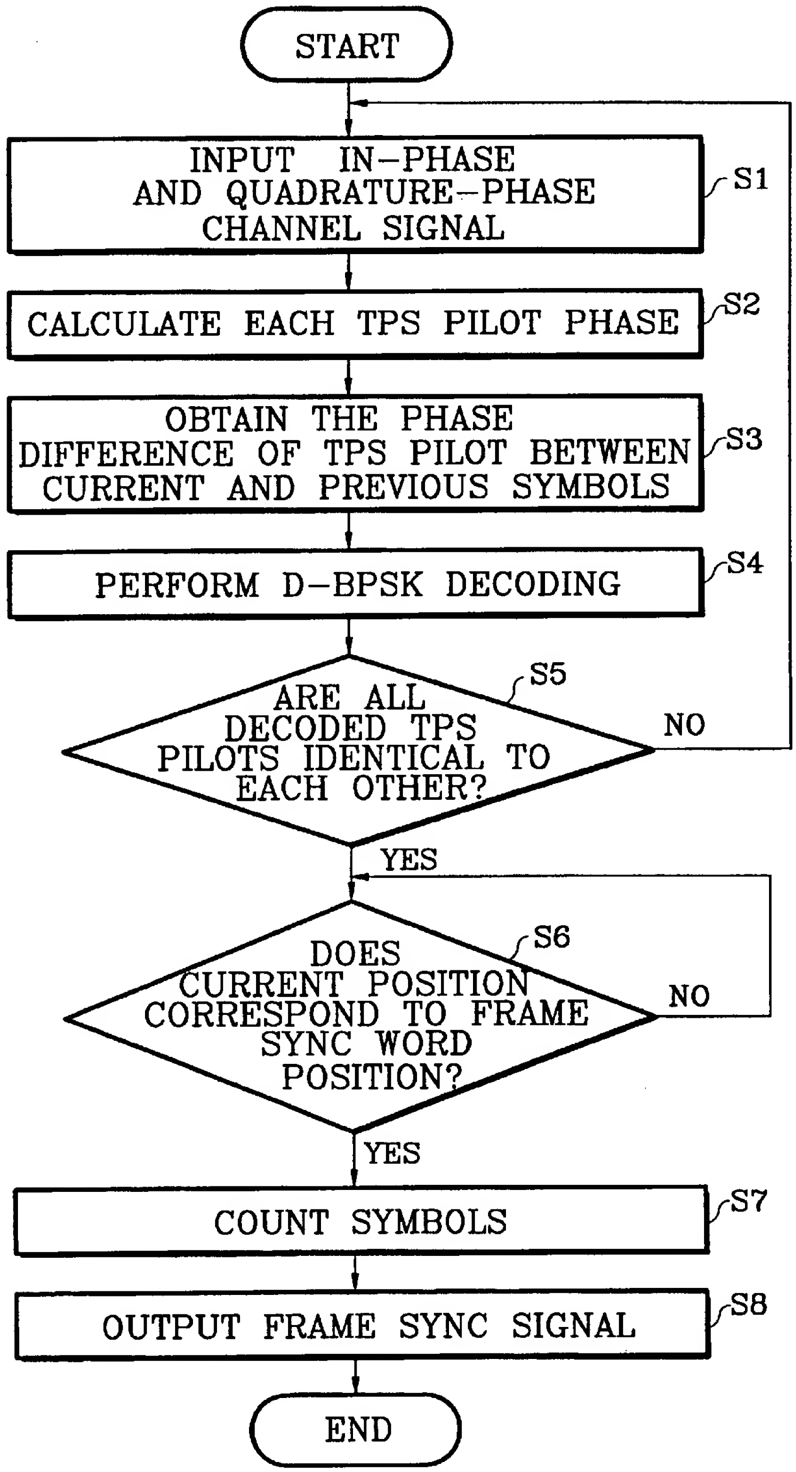

FRAME SYNC SIGNAL
400
430
COUNTER
SYMBOL CLOCK
410
420
S_{n-68}
S_n
0 1 . . . . . . 67 68
RESET
S_n
200
210
0 1 . . . . . . 16 17
0 1 2 3 4 5 6 7 8 9
$\theta_{n,p}$
10 bits
$\theta_{n-1,p}$
220
$\Delta\theta=\theta_{n,p}-\theta_{n-1,p}$
D-BPSK DECODER
230
$S_{n,p}$
300
310
0 1 . . . . 15 16
320
PILOT COMPARATOR
100
PHASE CALCULATOR
I
Q
10 bits

FIG.4


START
INPUT IN-PHASE AND QUADRATURE-PHASE CHANNEL SIGNAL
S1
CALCULATE EACH TPS PILOT PHASE
S2
OBTAIN THE PHASE DIFFERENCE OF TPS PILOT BETWEEN CURRENT AND PREVIOUS SYMBOLS
S3
PERFORM D-BPSK DECODING
S4
ARE ALL DECODED TPS PILOTS IDENTICAL TO EACH OTHER?
S5
NO
YES
DOES CURRENT POSITION CORRESPOND TO FRAME SYNC WORD POSITION?
S6
NO
YES
COUNT SYMBOLS
S7
OUTPUT FRAME SYNC SIGNAL
S8
END

FRAME SYNCHRONIZATION METHOD AND APPARATUS FOR USE IN DIGITAL COMMUNICATION SYSTEM UTILIZING OFDM METHOD

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to a receiver in a digital communication system utilizing an Orthogonal Frequency Division Multiplexing (OFDM) method, and more particularly to a method for performing frame synchronization by using characteristics of synchronization word inverted at each frame in a transmission parameter signaling (TPS) block, and an apparatus employing the same.

2. Description of the Prior Art

In a wireless communication channel and digital high-definition TV (HDTV) transmission channel, it is known that an Inter-Symbol Interference (ISI) caused by multi-path fading in a received signal commonly occurs. Particularly, when data for HDTV are transmitted through the channel at high speed, the ISI increases causing errors to be generated during the data recovery at the receiving side. To solve this problem, recently, OFDM method has been proposed as a transmission method for use in the Digital Audio Broadcasting (DAB) and Digital Terrestrial Television Broadcasting (DTTB) standards.

In OFDM method, serially-inputted symbol streams are divided into a predetermined unit block. The divided symbol streams of each unit block are converted into N number of parallel symbols. The N number of parallel symbols are multiplexed and added by using a plurality of subcarriers having different frequencies, respectively, according to Inverse Fast Fourier Transform (IFFT) algorithm. The added data are transmitted via the channel. That is, the N number of parallel symbols are defined as one unit block, and each subcarrier of the unit block has an orthogonal characteristic, which does not have an influence on subchannels. Compared to a conventional single carrier transmission method, OFDM method can reduce the ISI caused by the multi-path fading by maintaining the same symbol transmission rate and increasing symbol period as much as by the number of subchannels (N). Especially, in OFDM method, a guard interval (GI) is inserted between the transmitted symbols to enhance the capability of the ISI reduction, making it possible to realize a simplified structure of channel equalizer. In contrast to a conventional Frequency Division Multiplexing (FDM) type, OFDM method has a characteristic that spectrums of each subchannel are superimposed causing it to have a higher band efficiency. Further, the spectrum has a wave of rectangular shape and electric power is uniformly distributed at each frequency band, which prevents from being affected by the same channel interference. The OFDM method is commonly combined with modulation types such as Pulse Amplitude Modulation (PAM), Frequency Shift Keying (FSK), Phase Shift Keying (PSK), and Quadrature Amplitude Modulation (QAM).

FIGS. 1A to 1B are format diagrams of transmission symbol units of a conventional OFDM signal. Symbols transmitted from a transmitting side, as shown in FIG. 1A, comprises an useful part and a guard interval. The useful part contains useful OFDM samples, and the guard interval is inserted in front side of the useful part and separates OFDM samples into symbol units. Samples used in the guard interval are copies of samples located in lower portion of the useful part. According to DTTB standard, the size of the useful part is separated into 2K mode and 8K mode by a Fast

Fourier Transform (FFT) size. For 2K mode, as shown in FIG. 1B, the size of the useful part is defined by "2048" samples. In addition, the size of the guard interval is separated into ¼, ⅛, 1/16, and 1/32 of the FFT size. In case of ¼ of the FFT size, as shown in FIG. 1B, the size of the guard interval is defined by "512" samples. Here, "2048" is the sum of 1705 useful subcarriers and 343 NULL subcarriers. The guard interval is comprised of copied data from the last parts of the useful part, 1536-th data to 2047-th data (namely, 512 sizes). The guard interval is inserted in the front portion of the useful data. Finally, the size of transmission symbol units is defined by the sum (2560) of the useful part (2048) and the guard interval (512).

Meanwhile, according to DVB standard, an OFDM signal comprises frames having has 68 OFDM symbols, respectively and a super frame comprises four frames. Each frame comprises transmitted data, Continual Pilot Carriers (CPC), and a TPS pilot.

The transmitting side of the OFDM communication system performs IFFT for N number of symbols, defined as one block unit, and transmits it in frame units. The receiving side performs the FFT for the transmitted frame, to recover an original information. Accordingly, when the frames between the transmitting and receiving sides are not synchronized, errors are generated during the recovery of data.

SUMMARY OF THE INVENTION

In view of the foregoing, it is an object of the present invention to provide a frame synchronization method and apparatus for performing frame synchronization by using characteristics of synchronization word inverted at each frame in a transmission parameter signaling (TPS) block, in digital communication system utilizing OFDM method.

In order to achieve the above object, the present invention provides a frame synchronization method for use in a digital communication system utilizing OFDM method, comprising the steps of: a) calculating phase values of TPS pilots within one symbol according to in-phase and quadrature-phase channel signals received from a transmitting side; b) calculating respective phase differences from the phase values of the TPS pilots of previous symbol and the phase values of the TPS pilots of current symbol calculated in the step a); c) performing D-BPSK demodulation for the phase difference obtained in the step b); d) determining whether all the demodulated TPS pilots in the step c) are identical to each other; e) determining whether current position corresponds to a sync word position, when all the demodulated TPS pilots are determined identical to each other, in the step d); and f) counting symbols, when current position corresponds to the sync word position in said step e), and generating a frame sync signal according to the counted value.

In order to achieve the above object, the present invention provides a frame synchronization apparatus for use in a digital communication system utilizing OFDM method, comprising: phase calculation means for calculating phase values of TPS pilot within a symbol according to in-phase and quadrature-phase channel signal received from a transmitting side; D-BPSK demodulating means for performing D-BPSK demodulation for the phase values of TPS pilot supplied from the phase calculation means and outputting TPS pilots within the demodulated symbol; control signal generating means for comparing the demodulated TPS pilots with each other and outputting a control signal according to the comparison result; and frame synchronization means for confirming a sync word position according to the control signal supplied from the control signal generating means and outputting a frame sync signal.

BRIEF DESCRIPTION OF THE DRAWINGS

The above and other objects, features, and advantages of the present invention will be apparent from the following detailed description of the preferred embodiments of the invention in conjunction with the accompanying drawings, in which:

FIGS. 1A and 1B are format diagrams for a transmission symbol of a conventional OFDM signal;

FIG. 2 is a diagram illustrating a frame structure of an OFDM signal according to the present invention;

FIG. 3 is a block diagram illustrating a frame synchronization apparatus in a digital communication system utilizing OFDM method in accordance with a preferred embodiment of the present invention; and

FIG. 4 is a flowchart illustrating a frame synchronization method in a digital communication system utilizing OFDM method in accordance with the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Reference will now be made in detail to the present invention, examples of which are illustrated in the accompanying drawings. Wherever possible, the same reference numbers will be used throughout the drawings to refer to the same or like parts.

First, parameters according to two FFT size modes are represented by the following table 1.

TABLE 1

Parameter	8K mode	2K mode
number of subcarrier k	6817	1705
k_{min} subcarrier	0	0
k_{max} subcarrier	6816	1704
reciprocal number of subcarrier interval (Tu)	896 μs	224 μs
subcarrier interval (1/Tu)	1116 Hz	4464 Hz
interval between subcarriers, k_{min} and k_{max} {(k-1)/Tu}	7.61 Hz	7.61 Hz

That is, a symbol period Ts comprises the period Tu corresponding to the reciprocal of the subcarrier interval and a period ΔT corresponding to the guard interval.

Meanwhile, in accordance with an embodiment of the present invention, a frame synchronization between transmitting and receiving sides is performed by using a TPS pilot signal of various pilot signals. The TPS pilot signal is used to transmit an information related to the transmission, for example, a modulation information defined by α value of a QAM constellation pattern, a hierarchy information, a guard interval information, an inner code rate information, a frame number information, and etc., to the receiving side. 17 number of TPS pilots are used when the FFT size is 2K mode, whereas 68 number of TPS pilots are used when the FFT1 size is 8K mode. Subcarrier indexes for the TPS pilot are represented by the table 2.

TABLE 2

2K mode	8K mode
34 50 209 346 413 569 595 688 790 901 1073 1219 1262 1286 1469 1594 1687	34 50 209 346 413 569 595 688 790 901 1073 1219 1262 1286 1469 1594 1687 1738 1754 1913 2050 2117 2273 2299 2392 2494 2605 2777 2923 2966 2990 3173 3298 3391 3442 3458 3617 3754

TABLE 2-continued

2K mode	8K mode
	3821 3977 4003 4096 4198 4309 4481 4627 4670 4694 4877 5002 5095 5146 5162 5321 5458 5525 5681 5707 5800 5902 6013 6185 6331 6374 6398 6581 6706 6799

FIG. 2 shows a frame structure of the OFDM signal in accordance with the preferred embodiment of the present invention. Here, assume 2K mode, namely k_{min}=0 and k_{max}=1704. The subcarrier number of the TPS pilot, as shown in table 2, is 17 (TPS pilot #0~TPS pilot #16) within a symbol, and all TPS data within a symbol are the same. One frame comprises 68 symbols, and one TPS block for one frame contains TPS pilot of 68-bits.

Here, of one TPS block (68-bits), 1-bit is used for an initialization bit, 16-bits are used for synchronization bits, 37-bits are used for information bits, and 14-bits are used for redundancy bits for error protection. Of 37 information bits, 23-bits are used, the remaining 14-bits are reserved and set as "0". The TPS block is transmitted according to the following table 3.

TABLE 3

Symbol (bit) number	format	usage/content
S_0	0	initialization
S_1~S_{16}	0011010111101110 or 1100101000010001	synchronization word
S_{17}~S_{22}	011000	length indicator
S_{23}~S_{24}	Refer to Table 4	number of frame
S_{25}~S_{26}	Refer to Table 5	constellation
S_{27}~S_{29}	Refer to Table 6	hierarchy information
S_{30}~S_{32}	Refer to Table 7	code rate, HP stream
S_{33}~S_{35}	Refer to Table 7	code rate, LP stream
S_{36}~S_{37}	Refer to Table 8	guard interval
S_{38}~S_{39}	Refer to Table 9	transmission mode
S_{40}~S_{53}	All set to "0"	reserved
S_{54}~S_{57}	BCH code	error protection

Referring to table 3, the bit S_0 represents the initialization bit for Differential-Binary Phase Shift Keying (D-BPSK) demodulation. The 16-bits (S_1~S_{16}) are the synchronization words, and within each super frame, a first frame and a third frame have the synchronization word S_1~S_{16}="0011010111101110" and a second frame and a fourth frame have the synchronization word S_1~S_{16}="1100101000010001". Accordingly, in the embodiment of the present invention, frame synchronization between the transmitting and receiving sides is performed by using the characteristics that the synchronization word is inverted at each frame in TPS blocks. Meanwhile, each super frame contains four frames, and it is separated according to two bits S_{23} and S_{24}, like the following table 4.

TABLE 4

bit S_{23}, S_{24}	frame number
00	the first frame of super frame (0)
01	the second frame of super frame (1)
10	the third frame of super frame (2)
11	the fourth frame of super frame (3)

The bits S_{25} and S_{26} represent constellation characteristics shown by the following table 5.

TABLE 5

bits S_{25}, S_{26}	constellation characteristic
00	QPSK
01	16-QAM
10	64-QAM
11	reserved bit

The bits S_{27}, S_{28}, and S_{29} represent hierarchy information shown by the following table 6.

TABLE 6

bits S_{27}, S_{28}, S_{29}	α value
000	non-hierarchy
001	$\alpha = 1$
010	$\alpha = 2$
011	$\alpha = 4$
100	reserved
101	reserved
110	reserved
111	reserved

Namely, the hierarchy information indicates whether or not the transmission is hierarchical, having α value if it is hierarchical.

Non-hierarchy channel coding and modulation requires a signal corresponding to a code rate. Here, three bits for determining the code rate are represented by the following table 7.

TABLE 7

bits S_{30}, S_{31}, S_{32} (HP stream) bits S_{33}, S_{34}, S_{35} (HP stream)	code rate
000	1/2
001	2/3
010	3/4
011	5/6
100	7/8
101	reserved
110	reserved
111	reserved

The bits S_{36} and S_{37} represent the size of the guard interval shown by the following table 8. In the embodiment of the present invention, assume S_{36}, S_{37}="11", namely ¼.

TABLE 8

bits S_{36}, S_{37}	the size of the guard interval (Δ/Tu)
00	1/32
01	1/16
10	1/8
11	1/4

The bits S_{38} and S_{39} represent transmission modes shown by the following table 9. In the embodiment of the present invention, assume S_{38}, S_{39}"00", namely 2K mode.

TABLE 9

Bits S_{38}, S_{39}	transmission mode
00	2K mode
01	8K mode
10	reserved
11	reserved

FIG. 3 is a block diagram illustrating a frame synchronization apparatus in a digital communication system utilizing OFDM method in accordance with a preferred embodiment of the present invention. The frame synchronization apparatus comprises a phase calculator **100**, a Differential-Binary Phase Shifted Keying (D-BPSK) demodulator **200**, a control signal generator **300**, and a frame synchronization unit **400**. The D-BPSK demodulator **200** comprises a phase storage unit **210**, a subtractor **220**, and a D-BPSK decoder **230**. The control signal generator **300** comprises a pilot storage unit **310** and a pilot comparator **320**. The frame synchronization unit **400** comprises a TPS pilot storage unit **410**, a TPS pilot comparator **420**, and a counter **430**. The symbol $\Theta_{n,p}$ denotes a phase of p-th TPS pilot of current n-th symbol and the symbol $\Theta_{n-1,p}$ denotes a phase of p-th TPS pilot of previous (n−1)th symbol. Further, the symbol $\Delta\Theta_n$ denotes the phase difference of p-th TPS pilot between current n-th symbol and previous (n−1)th symbol; $S_{n,p}$ denotes a D-BPSK decoded TPS bit for p-th TPS pilot of current n-th symbol; and S_n denotes a TPS bit of current n-th symbol.

Referring to FIG. **3**, the phase calculator **100** receives in-phase and quadrature-phase channel signals from the transmitting side and calculates the phase $\Theta_{n,p}$ of p-th TPS pilot of current n-th symbol, where p ranges from 1 to 17. At this time, the calculated phase $\Theta_{n,p}$ is stored in a built-in memory, for example, Read Only Memory (ROM) in the form of a look-up table, in advance.

The D-BPSK demodulator **200** performs the D-BPSK demodulation for the phase $\Theta_{n,p}$ of TPS pilot outputted from the phase calculator **100** and outputs the TPS pilot within the demodulated symbol. That is, the phase $\Theta_{n,p}$ of TPS pilot outputted from the phase calculator **100** is stored in the phase storage unit **210**. The subtracter **220** subtracts the phase $\Theta_{n-1,p}$ of p-th TPS pilot of previous (n−1)th symbol from the phase $\Theta_{n,p}$ of p-th TPS pilot of current n-th symbol supplied from the phase storage unit **210** and outputs a phase difference $\Delta\Theta_n$. The D-BPSK decoder **230** performs the D-BPSK decoding for the phase difference $\Delta\Theta_n$ supplied from the subtracter **220** and outputs a decoded TPS pilot $S_{n,p}$. Here, the phase storage unit **210** can be implemented by the shift register capable of storing 18 phases, being one more than the corresponding 17 pilots within a symbol. The phase of each TPS pilot is stored in the shift register in unit of 10-bits.

The control signal generator **300** compares the decoded TPS pilots $S_{n,p}$ supplied from the D-BPSK demodulator **200** with each other and outputs a control signal according to the compared result. That is, 17 number of the decoded TPS pilot $S_{n,p}$ outputted from the D-BPSK demodulator **200** are stored in the pilot storage unit **310**. The pilot comparator **320** compares 17 number of the decoded TPS pilots $S_{n,p}$ with each other and determines whether all the decoded TPS pilots $S_{n,p}$ are identical to each other. If all the decoded TPS pilots $S_{n,p}$ are identical to each other, the TPS pilot S_n of corresponding symbol is outputted. Otherwise, the reset signal for resetting the frame synchronization unit **400** is outputted. Here, the pilot storage unit **310** can be implemented by the shift register capable of storing 17 pilots within a symbol.

The frame synchronization unit **400** confirms the sync word position (refer to the above table 3) converted at each frame according to the control signal supplied from the control signal generator **300** and outputs a frame sync signal. That is, the TPS pilot of the corresponding n-th symbol outputted from the control signal generator **300** is stored in the TPS pilot storage unit **410**. The TPS pilot comparator **420** compares the TPS pilot S_{n-68} of previous frame with the TPS pilot S_n of current frame, confirms the sync word position according to the compared result, and outputs a control signal when the current position corresponds to the sync word position. That is, the TPS pilot comparator **420** outputs "0", which means a sync word position, when the TPS pilot S_{n-68} of previous frame is identical to the TPS pilot S_n of current frame. Otherwise, the TPS pilot comparator **420** outputs "1". The counter **430** counts symbol clocks according to the control signal outputted from the TPS pilot comparator **420** and outputs a frame sync signal. That is, the counter **430** counts the symbol clocks when the TPS pilot comparator **420** outputs "0" and finally outputs the counted value "16". Here, the TPS pilot storage unit **410** can be implemented by a shift register capable of storing 69 TPS pilots, being one more than corresponding 68 symbols within a frame. Also, the TPS pilot comparator **420** can be implemented by an exclusive OR gate and the counter **430** can be implemented by a 4 bits counter to count the sync word of 16 bits (refer to the above table 3) converted at each frame.

FIG. **4** is a flowchart illustrating a frame synchronization method in a digital communication system utilizing OFDM method in accordance with the present invention.

Referring to FIG. **4**, in the step **S1**, in-phase and quadrature-phase channel signals are inputted. In the step **S2**, the phase of the TPS pilot according to the in-phase and quadrature-phase channel signal inputted in the step **S1** is calculated. In the step **S3**, the phase difference is obtained from the phase of the TPS pilot of previous symbol and the phase of the TPS pilot of current symbol calculated in the step S2. In the step **S4**, the phase difference obtained in the step S3 is demodulated by D-BPSK. In the step **S5**, it is determined whether all the decoded TPS pilots $S_{n,p}$ are identical to each other. If all the decoded TPS pilots $S_{n,p}$ are not identical to each other, the process returns to the step **S1**. Otherwise, it is determined whether the current position corresponds to the sync word position in the step **S6**. In the step S7, when the current position corresponds to the sync word position in the step **S6**, the number of symbols is counted and the frame sync signal is outputted in the step **S8**.

In the embodiment of the present invention, the operation of the present invention has been described with regard to the case of the 2K FFT size mode. Additionally, the application of the embodiment can be applied to the 8K FTT size mode.

As described above, the frame synchronization method and apparatus of the present invention can perform frame synchronization by using the synchronization word inverted at each frame in one TPS block without the need to increase its hardware.

While this invention has been described in connection with what is presently considered to be the most practical and preferred embodiments, it is to be understood that the invention is not limited to the disclosed embodiment, but, on the contrary, it is intended to cover various modifications and equivalent arrangements included within the spirit and scope of the appended claims.

What is claimed is:

1. A frame synchronization method for use in a digital communication system utilizing OFDM method, comprising the steps of:

a) calculating phase values of TPS pilots within a symbol according to in-phase and quadrature-phase channel signals received from a transmitting side;

b) calculating phase differences from the phase values of the TPS pilots of previous symbol and the respective phase values of the TPS pilots of current symbol calculated in said step a);

c) performing D-BPSK demodulation for the phase difference obtained in said step b);

d) determining whether all the demodulated TPS pilots in said step c) are identical to each other;

e) determining whether current position corresponds to a sync word position, when all the demodulated TPS pilots are determined identical to each other, in said step d);

and

f) counting symbols, when current position corresponds to the sync word position in said step e), and generating a frame sync signal according to the counted value.

2. A frame synchronization apparatus for use in a digital communication system utilizing OFDM method, comprising:

phase calculation means for calculating phase values of TPS pilots within a symbol according to in-phase and quadrature-phase channel signals received from a transmitting side;

D-BPSK demodulating means for performing D-BPSK demodulation for the phase values of the TPS pilots supplied from said phase calculation means and outputting the TPS pilots within the demodulated symbol;

control signal generating means for comparing the demodulated TPS pilots with each other and outputting a control signal according to the compared result; and

frame synchronization means for confirming a sync word position according to the control signal supplied from said control signal generating means and outputting a frame sync signal.

3. The frame synchronization apparatus of claim **2**, wherein said phase calculating means is implemented by a memory in which the phase values of the TPS pilots corresponding to the in-phase and quadrature-phase channel signals are stored in the form of a look-up table.

4. The frame synchronization apparatus of claim **2**, wherein said D-BPSK demodulating means comprises;

a phase storage unit for storing the phase values of the TPS pilots outputted from said phase calculating means;

a subtracter for subtracting the phase values of the TPS pilots of previous symbol from the respective phase values of the TPS pilots of current symbol supplied from said phase storage unit and outputting phase differences; and

a D-BPSK decoder for performing D-BPSK decoding for the phase difference supplied from said subtracter and outputting a decoded TPS pilot.

5. The frame synchronization apparatus of claim **4**, wherein said phase storage unit is implemented by a shift register for storing number of phases being one more than corresponding number of pilots within one symbol.

6. The frame synchronization apparatus of claim **2**, wherein said control signal generating means comprises:

a pilot storage unit for storing the decoded TPS pilots outputted from said D-BPSK demodulating means; and

a pilot comparator for comparing the decoded TPS pilots with each other to determine whether all the decoded TPS pilots are identical to each other and generating a control signal according to the compared result.

7. The frame synchronization apparatus of claim 6, wherein the control signal outputted from said pilot comparator resets said frame synchronization means when all the decoded TPS pilots are not identical to each other.

8. The frame synchronization apparatus of claim 6, wherein said pilot storage unit is implemented by a shift register for storing number of pilots within one symbol.

9. The frame synchronization apparatus of claim 2, wherein said frame synchronization means comprises:

a TPS pilot storage unit for storing the TPS pilot of the corresponding symbol outputted from said pilot comparing means;

a TPS pilot comparator for comparing the TPS pilots of previous frame with the respective TPS pilots of current frame, confirming the sync word position according to the compared result, and outputting a control signal when the current position corresponds to the sync word position; and

counting means for counting symbol clocks according the control signal outputted from said TPS pilot comparator and outputting a frame sync signal according to the counted value.

10. The frame synchronization apparatus of claim 9, wherein said TPS pilot storage unit is implemented by a shift register for storing number of TPS pilots being one more than corresponding number of symbols within one frame.

11. The frame synchronization apparatus of claim 9, wherein said TPS pilot comparator is implemented by an exclusive OR gate.

12. The frame synchronization apparatus of claim 9, wherein said counting means is implemented by a 4-bit counter.

* * * * *

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Cupo et al.

(10) **Patent No.: US 6,347,071 B1**
(45) **Date of Patent: Feb. 12, 2002**

(54) **TIME DIVISION MULTIPLEXED TRANSMISSION OF OFDM SYMBOLS**

(75) Inventors: **Robert Louis Cupo**, Eatontown; **Mohsen Sarraf**, Rumson; **Mojtaba Shariat**, Matawan; **Mohammad Hossein Zarrabizadeh**, Woodbridge, all of NJ (US)

(73) Assignee: **Lucent Technologies Inc.**, Murray Hill, NJ (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

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(51) **Int. Cl.**[7] **H04J 11/00**
(52) **U.S. Cl.** **370/203**; 370/537
(58) **Field of Search** 370/203, 204–206, 370/208–209, 480, 532, 537, 540

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Primary Examiner—David R. Vincent
(74) *Attorney, Agent, or Firm*—William H. Bollman

(57) **ABSTRACT**

An orthogonal frequency division multiplexing (OFDM) technique which is time division multiplexed to reduce the overall effect on individual services from conditions such as selective fading. In accordance with the principles of the present invention, all available subcarriers in a channel are assigned to fewer than all of the requesting services, e.g., to just one particular service for a period of time. The period of time is preferably independent of the length of a conventional data frame. Thereafter, a second service is assigned access to the use of all available subcarriers for a period of time corresponding to its required bandwidth, and so on until all requesting services are allotted a portion of time for access to all available subcarriers. Any one service may utilize any number of the available subcarriers in a particular superframe containing one cycle of transmissions for all services.

7 Claims, 9 Drawing Sheets

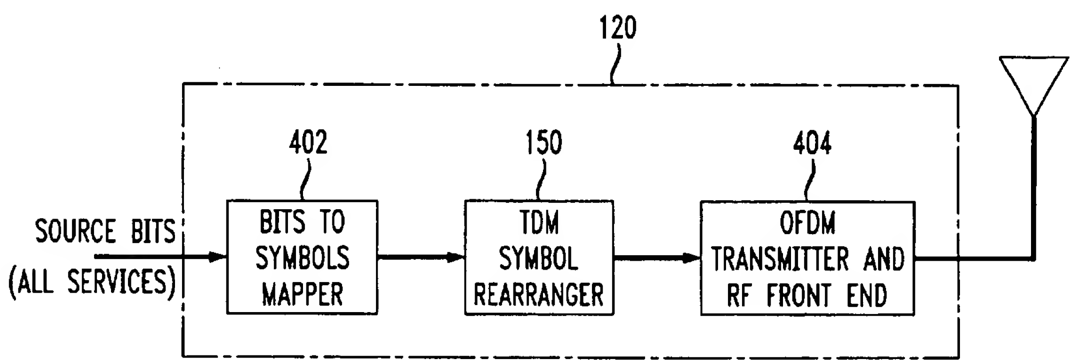


FIG. 1

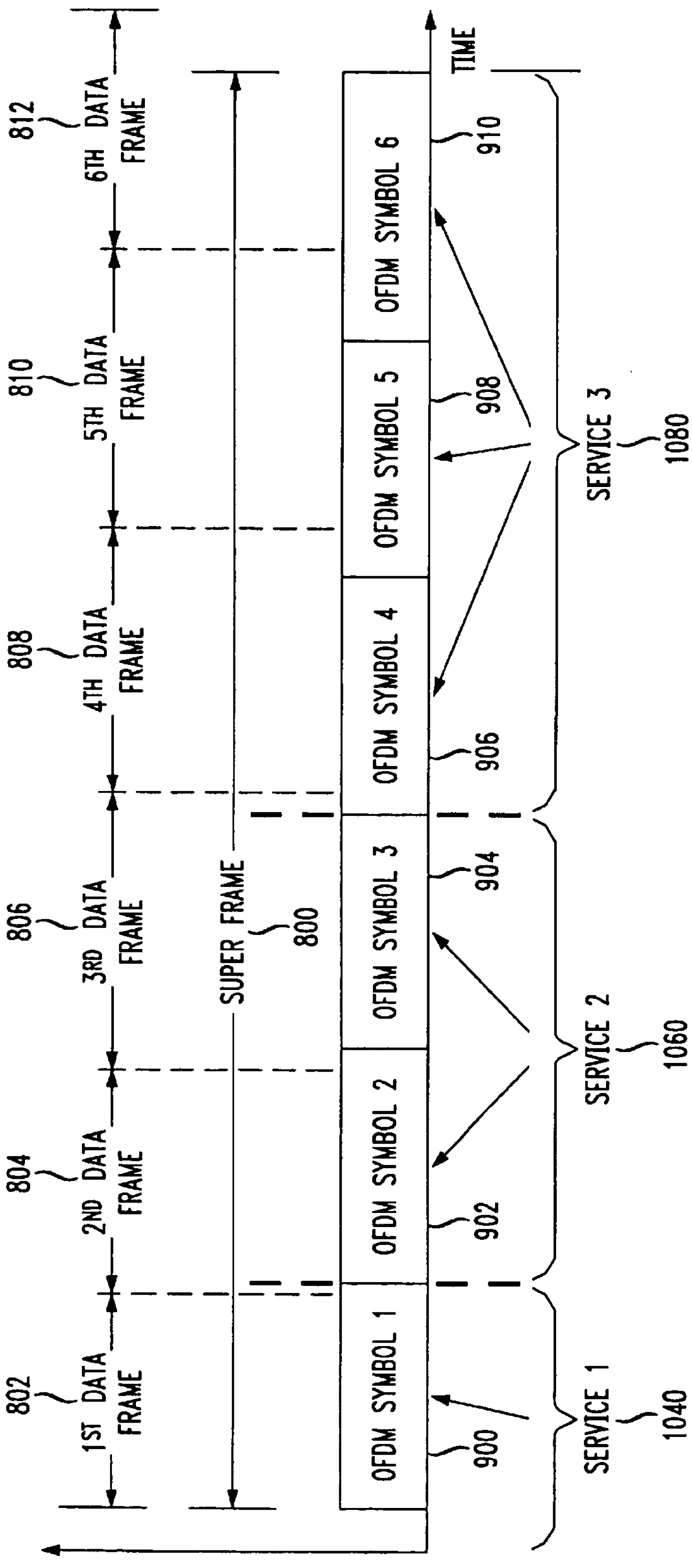

TIME
812
6TH DATA FRAME
810
5TH DATA FRAME
808
4TH DATA FRAME
806
3RD DATA FRAME
804
2ND DATA FRAME
802
1ST DATA FRAME
SUPER FRAME
800
OFDM SYMBOL 6
OFDM SYMBOL 5
OFDM SYMBOL 4
OFDM SYMBOL 3
OFDM SYMBOL 2
OFDM SYMBOL 1
910
908
906
904
902
900
SERVICE 3
1080
SERVICE 2
1060
SERVICE 1
1040

FIG. 2A

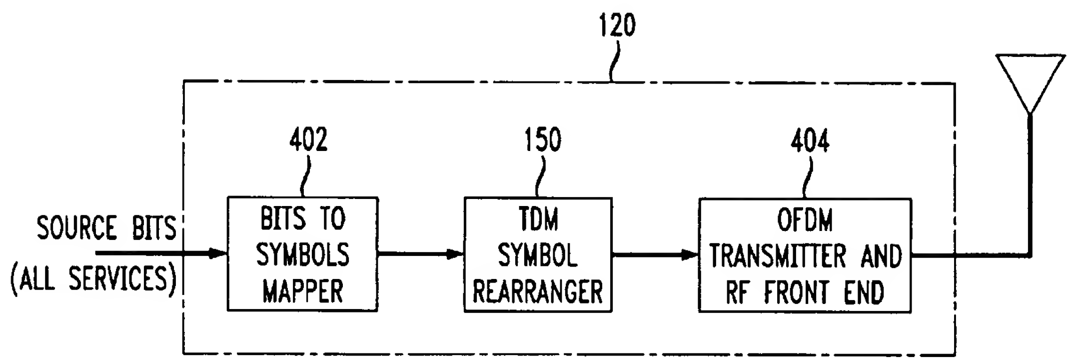

120
402
150
404
SOURCE BITS
(ALL SERVICES)
BITS TO SYMBOLS MAPPER
TDM SYMBOL REARRANGER
OFDM TRANSMITTER AND RF FRONT END

FIG. 2B

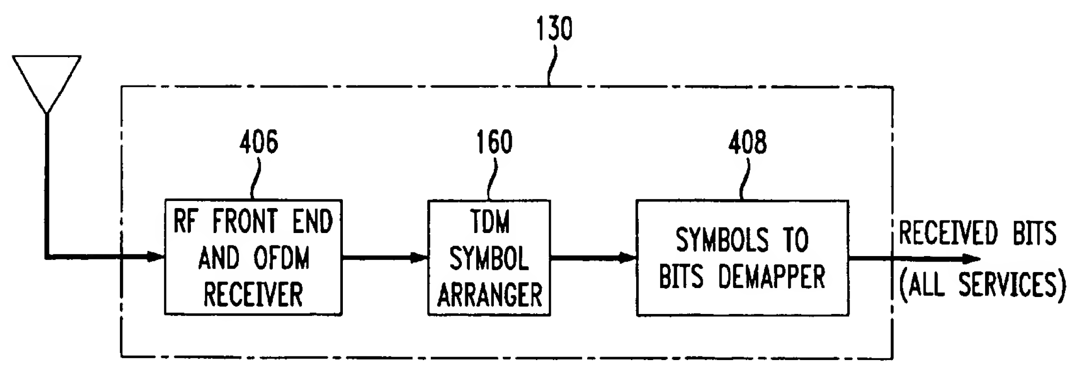

130
406
160
408
RF FRONT END AND OFDM RECEIVER
TDM SYMBOL ARRANGER
SYMBOLS TO BITS DEMAPPER
RECEIVED BITS
(ALL SERVICES)

FIG. 3A

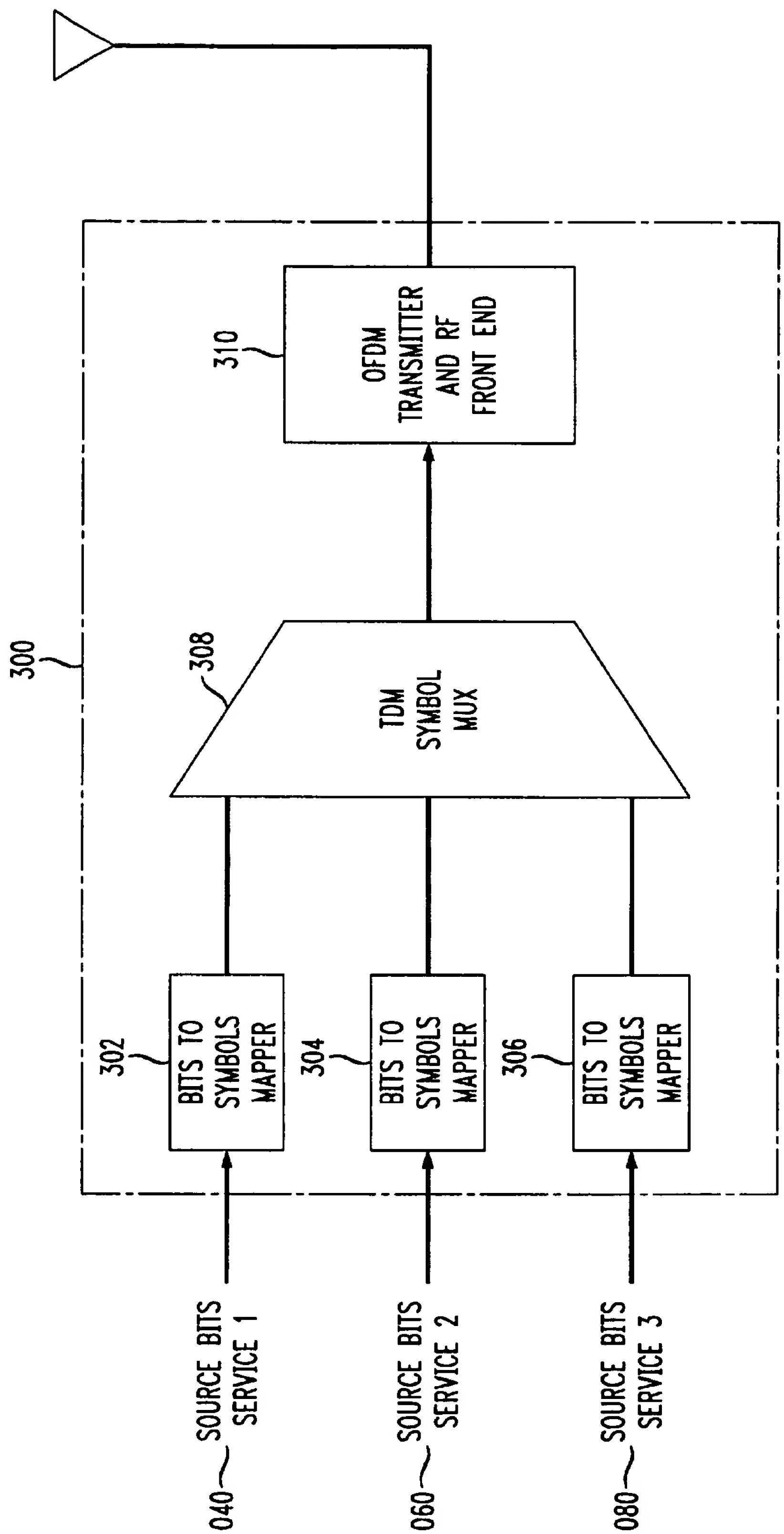
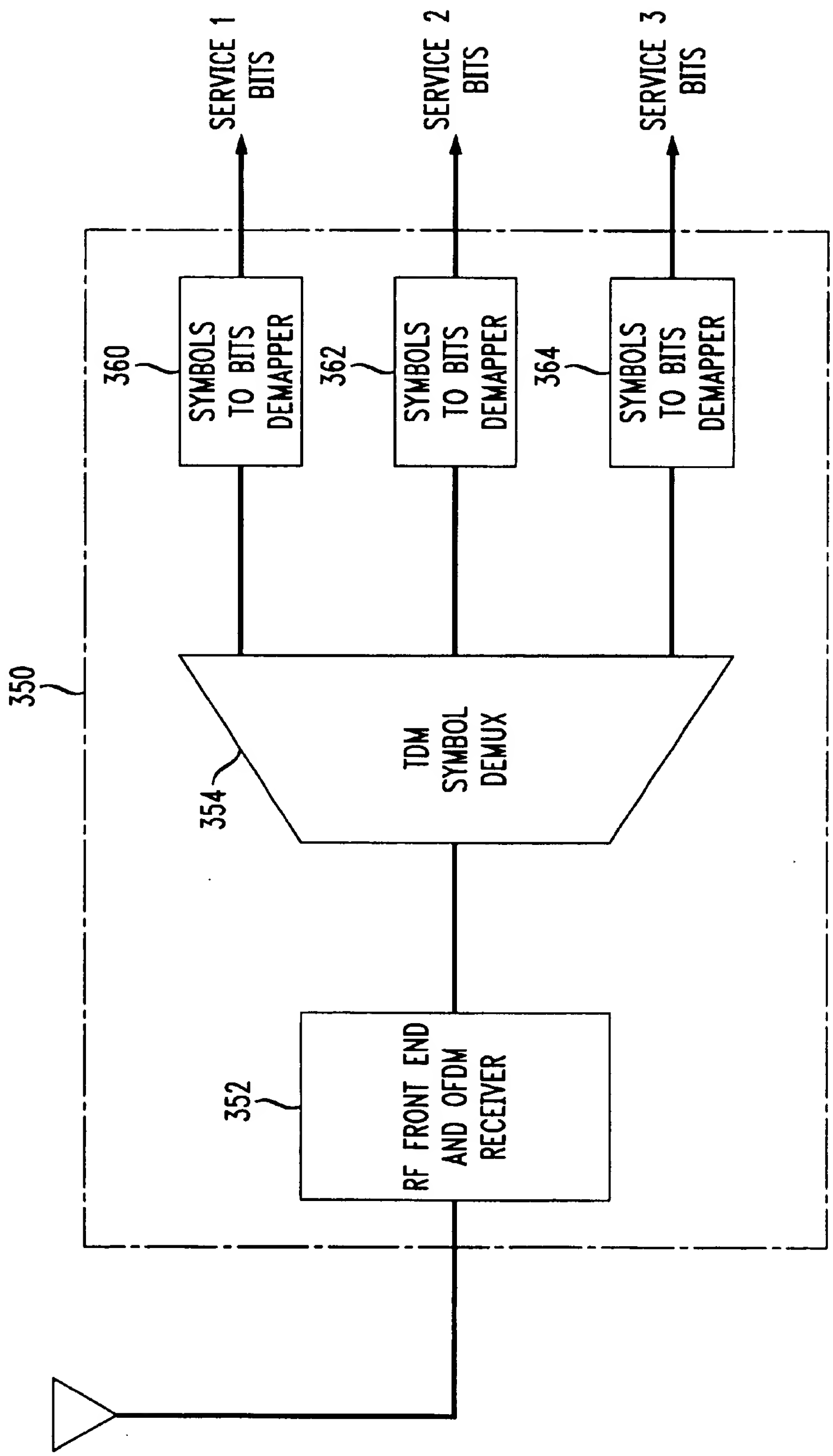

300
310
OFDM TRANSMITTER AND RF FRONT END
308
TDM SYMBOL MUX
302
BITS TO SYMBOLS MAPPER
304
BITS TO SYMBOLS MAPPER
306
BITS TO SYMBOLS MAPPER
1040
SOURCE BITS SERVICE 1
1060
SOURCE BITS SERVICE 2
1080
SOURCE BITS SERVICE 3

FIG. 3B


SERVICE 1 BITS
SERVICE 2 BITS
SERVICE 3 BITS
360
SYMBOLS TO BITS DEMAPPER
362
SYMBOLS TO BITS DEMAPPER
364
SYMBOLS TO BITS DEMAPPER
350
354
TDM SYMBOL DEMUX
352
RF FRONT END AND OFDM RECEIVER

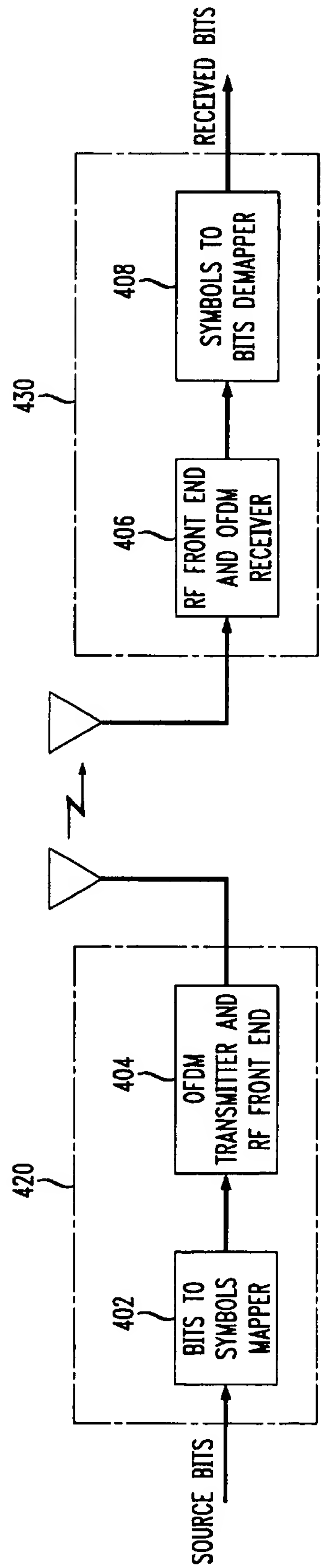


FIG. 4
PRIOR ART

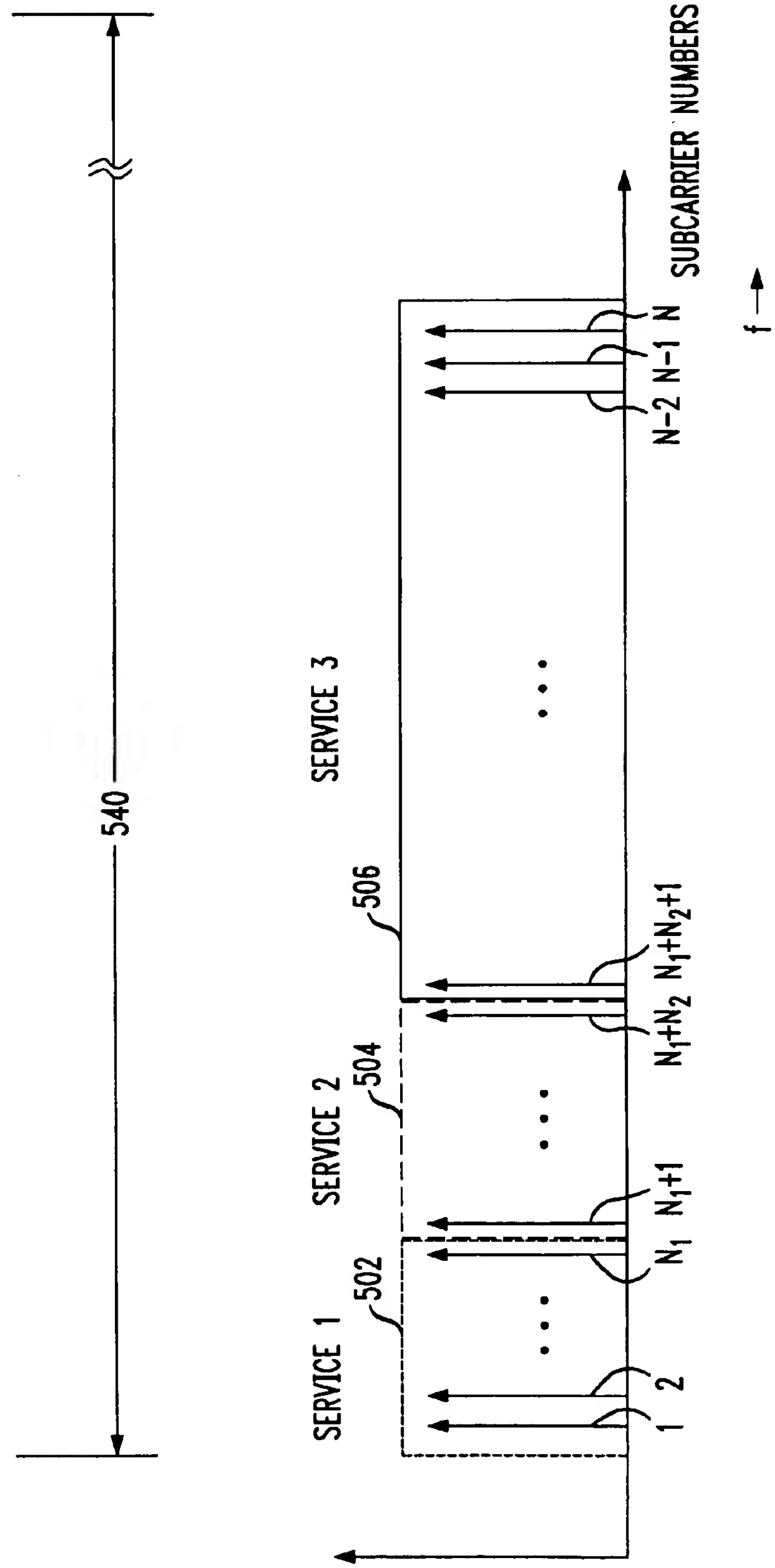


FIG. 5
PRIOR ART

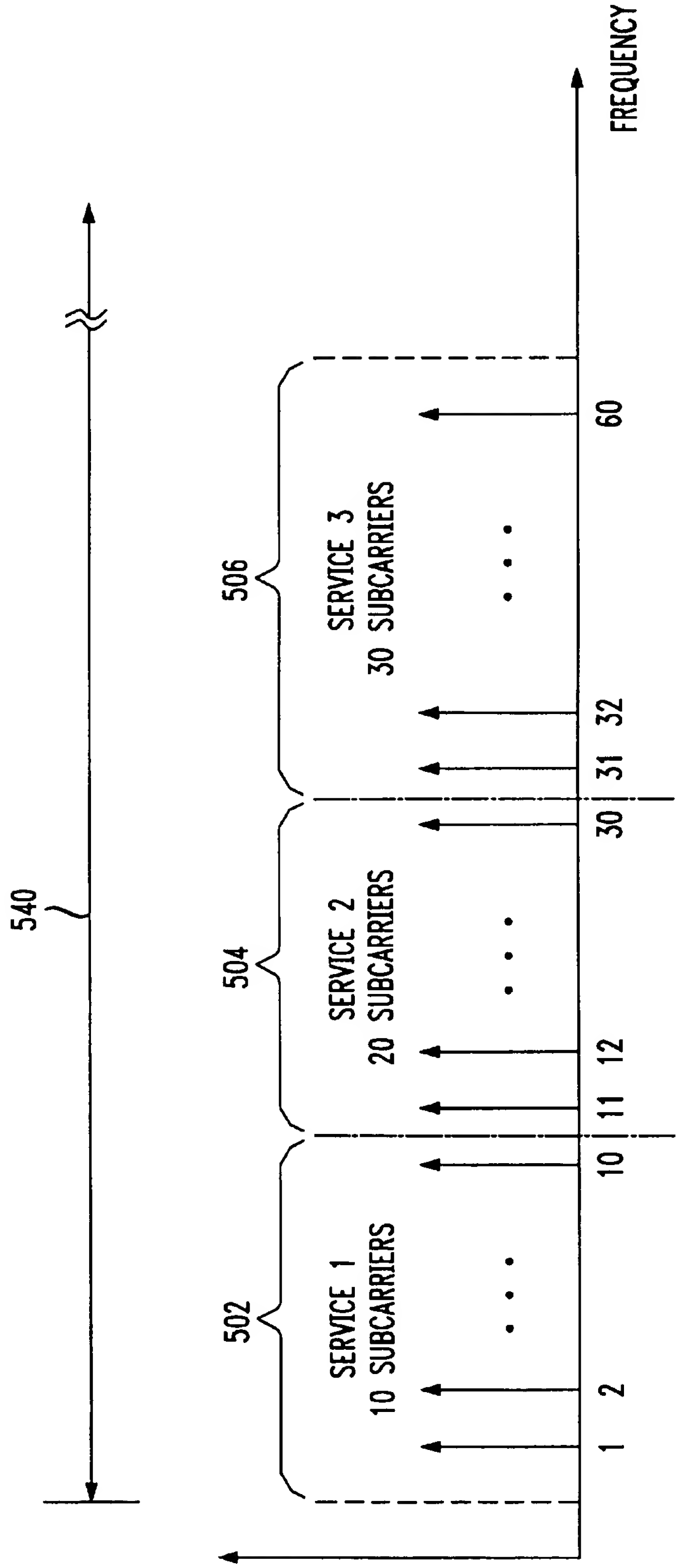
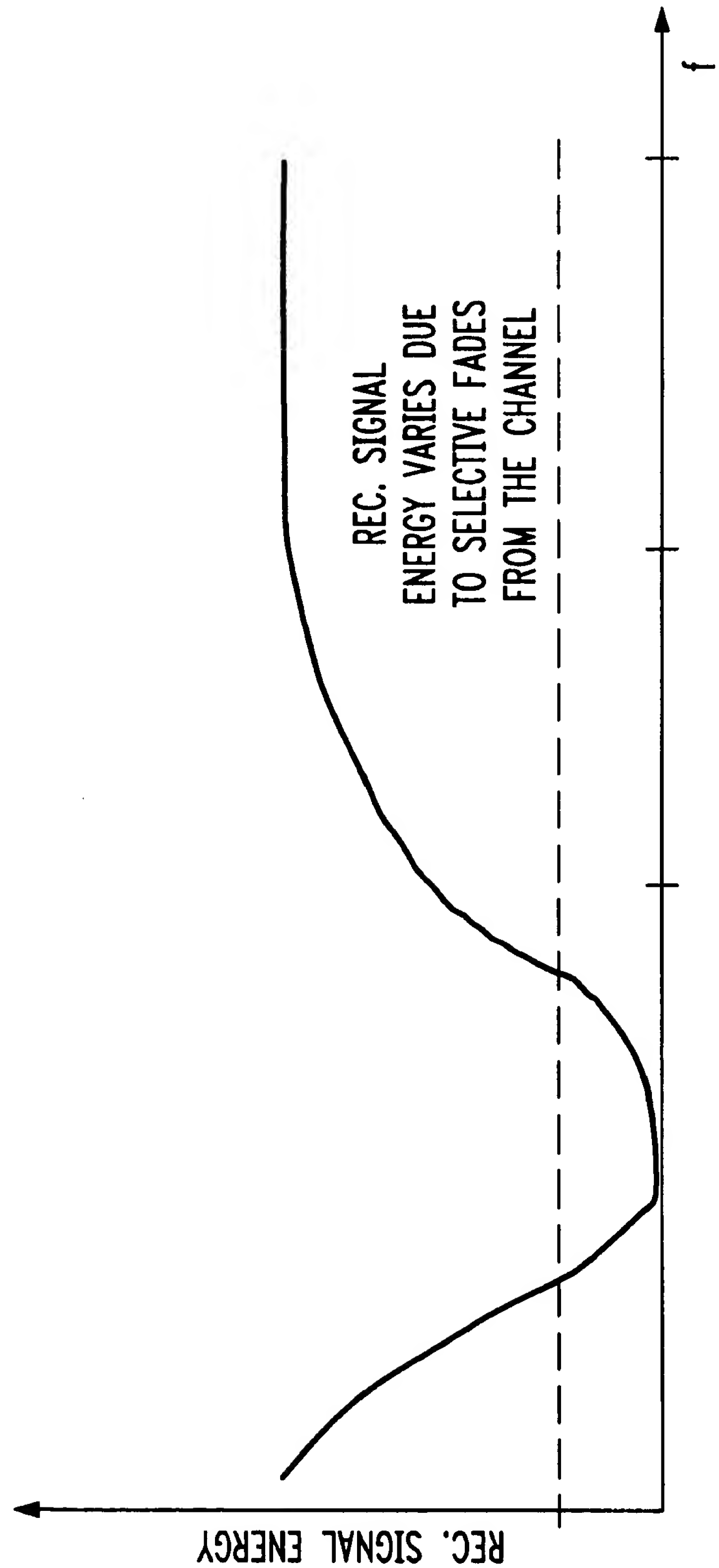


FIG. 6
PRIOR ART

FIG. 7A
PRIOR ART



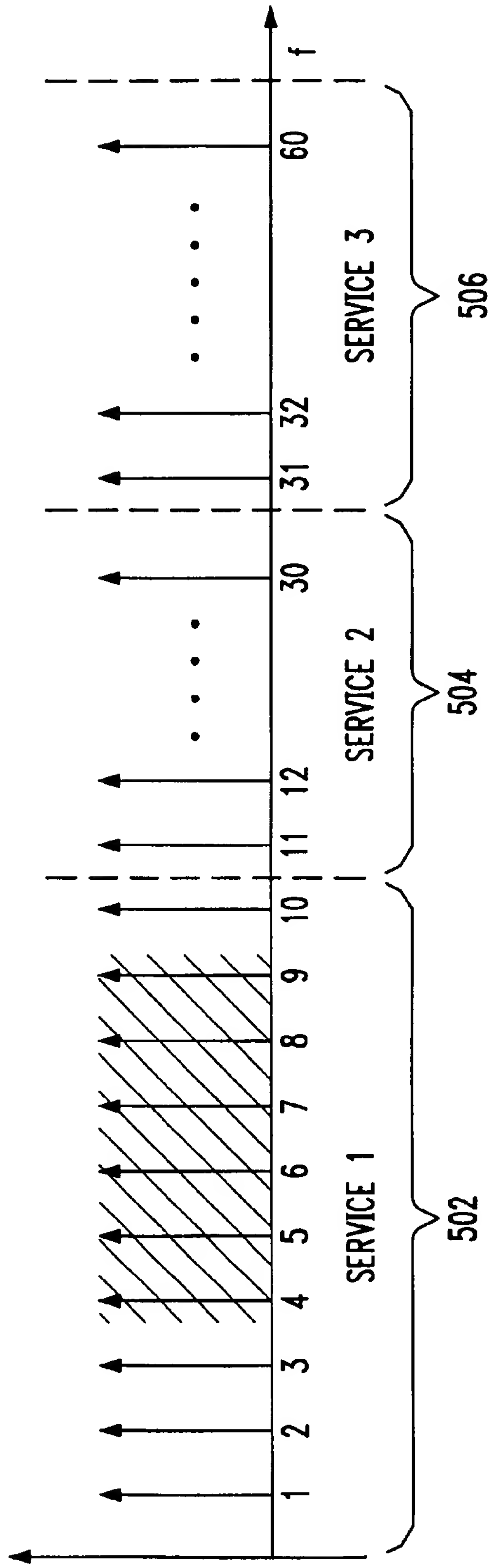


FIG. 7B
PRIOR ART

TIME DIVISION MULTIPLEXED TRANSMISSION OF OFDM SYMBOLS

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates generally to any system using a robust orthogonal frequency division multiplexing (OFDM) transmission scheme which is capable of reliably carrying a plurality of services or programs (referred to collectively herein as services) within each channel of allocated bandwidth even in a selective channel fading environment.

2. Background of Related Art

Orthogonal frequency division multiplexing (OFDM) is a conventional technique for transmitting data symbols using mutually independent and separated radio frequency (RF) subcarriers. OFDM has several desirable properties, e.g., it simplifies or even eliminates equalization problems considerably, has graceful performance degradation, and because of the absence of equalization, can be lower in complexity.

In a traditional multi-service OFDM system respective groups of available subcarriers in a given channel are assigned to each supported service according to the service's bandwidth needs. Therefore, OFDM symbols typically comprise two or more services each.

FIG. 4 shows a block diagram of a conventional OFDM transmission system.

In particular, in the transmitting portion **420** of the OFDM transmission system shown in FIG. **4**, a data source containing source bits from all the different services or programs (e.g., three different digital audio broadcast (DAB) services or programs) is input to a bits to symbol mapper **402**. The bits to symbol mapper **402** maps the data bits for current data output from respective DAB services or programs into a contiguous symbol stream. The contiguous symbol stream is input to an OFDM transmitter and radio frequency (RF) transmitter front end **404**, which transmits the contiguous symbol stream using the assigned portion of the available OFDM subcarrier frequencies.

At the receiving portion **430** of the OFDM transmission system shown in FIG. **4**, an RF receiver front end and OFDM receiver **406** receives the contiguous symbol stream containing the information for the number of different DAB services or programs (e.g., for three different DAB services or programs). A symbols to bits demapper **408** converts the contiguous symbol stream back into a data bit stream containing data for all of the different DAB services or programs.

FIG. **5** shows a conventional frequency distribution of OFDM subcarriers **1** to N for use by the different DAB services or programs, e.g., three different DAB services or programs **502–506**. The OFDM subcarriers **1** to N represent in this example all of the available subcarriers used in the transmission between the transmitting portion **420** and receiving portion **430** of a conventional OFDM transmission system, e.g., as shown in FIG. **4**.

When a number of different DAB services or programs **502–506**, e.g., three, are simultaneously transmitted, the available OFDM subcarriers **1** to N are conventionally distributed in frequency among the plurality of services **502–506**. For instance, the available OFDM subcarriers **1** to N are typically assigned in fixed, contiguous groups with respect to frequency between the three different DAB services or programs **502–506** as depicted in FIG. **5**. The number of subcarriers assigned to each DAB service or program is application specific, and typically depends upon the information capacity required by each DAB service or program.

Thus, in the disclosed example, a first DAB service or program **502** is assigned the first contiguous group N_1 of all available OFDM subcarriers (e.g., **1** to N_1), a second DAB service **504** is assigned the next contiguous group N_2 of the remaining available OFDM subcarriers (e.g., N_1+1 to N_1+N_2), and the third DAB service or program **506** is assigned the last contiguous group N_3 of all available OFDM subcarriers (e.g., N_1+N_2+1 to N).

However, a transmission channel **540** (e.g., an FM station) containing the OFDM subcarriers **1** to N of the different DAB services or programs **502–506** may be subject to selective fading affecting some of the OFDM subcarriers but not others. This is particularly true in a fixed or slow speed mobile environment. In such a case, a large number of the assigned subcarriers of one DAB service or program may be detrimentally affected while at the same time all or most of the assigned subcarriers of other DAB services or programs may be unaffected.

This example is further illustrated in FIGS. **6** and **7** demonstrating the effect of selective fading on some but not all OFDM subcarriers.

In particular, in FIG. **6**, at least sixty subcarrier frequencies **1** to **60** are assumed to be available in each data frame in the frequency domain. In the disclosed example, the first service **502** has the first ten (10) OFDM subcarrier frequencies **1** to **10** assigned thereto, the second service **504** has the next twenty (20) subcarriers **11** to **30** assigned thereto, and the third service **506** has the last thirty (30) subcarriers **31** to **60** assigned thereto.

Each of the different DAB services or programs **502–506** may conventionally have its own interleaver and/or Forward Error Correction (FEC) scheme to improve the quality of the transmission channel **540**.

FIGS. 7A and 7B show a relevant portion of a possible fading scenario in the example shown in FIG. **6** in which five of the subcarriers of the first service **502** are hit by a frequency selective fade. In such a case, the fourth through ninth subcarriers **4** to **9** assigned to the first service **502** are shown as detrimentally affected by the selective frequency fade and likely lost. This loss of ½(i.e., five (5) out of the ten (10) subcarriers or 50%) of the total subcarriers assigned to the first service **502** may be beyond the error recovery capability of the FEC scheme used for that particular service. In this example, however, the other two services **504** and **506** do not suffer from the frequency selective fades. Thus, while the symbols transmitted by the first service **502** during that use or data frame of the available subcarriers would likely be lost, the symbols transmitted by the other services **504** and **506** would be unaffected.

While the interleaver and/or FEC function of the various services or programs may be suitable to maintain reliable communications in the transmission channel **540**, e.g., an FM station in the absence of selective fading, the interleaver and FEC function of the one service or program may not be adequate to fully overcome the deterioration of the data communication due to channel fades, particularly when they affect a large percentage of the assigned subcarrier frequencies for just one service or a small percentage of a large number of requesting services for the channel. Thus, the impact of channel fades may affect some services more than other services.

Accordingly, in a frequency selective environment, one or more of the services may be rendered useless at the receiver

due to channel impairments. There is thus a need to improve the reliability of all or substantially all services or programs transmitted using OFDM modulation techniques within a transmission channel, e.g., within an FM channel.

SUMMARY OF THE INVENTION

In accordance with the principles of the present invention, a time division multiplexed orthogonal frequency division multiplexed transmitter comprises a bits to symbols mapper, a time division multiplex symbol rearranger in communication with the bits to symbols mapper, and an OFDM transmitter and radio frequency front end in communication with the time division multiplex symbol rearranger.

In another aspect of the present invention, a time division multiplexed orthogonal frequency division multiplexed receiver comprises a radio frequency front end and OFDM receiver, a time division multiplex symbol arranger in communication with the radio frequency front end and OFDM receiver, and a symbols to bits demapper in communication with the time division multiplex symbol arranger.

A method of transmitting symbols relating to a plurality of services in accordance with the principles of the present invention comprises assigning respective periods of time for transmission to each of a plurality of services. At least one symbol is firstly transmitted for a first one of the plurality of services using substantially all available subcarrier frequencies in an orthogonal frequency division multiplexed transmission system for a first period of time assigned to the first one of the plurality of services.

A method of transmitting symbols relating to a plurality of services in accordance with another aspect of the present invention comprises assigning respective periods of time for transmission to each of a plurality of services, and transmitting data from less than all of the plurality of services in a data frame corresponding to a use of all of a plurality of available subcarrier frequencies.

BRIEF DESCRIPTION OF THE DRAWINGS

Features and advantages of the present invention will become apparent to those skilled in the art from the following description with reference to the drawings, in which:

FIG. 1 shows a time division multiplexed (TDM) orthogonal frequency division multiplexing (OFDM) superframe in an exemplary OFDM transmission in accordance with the principles of the present invention.

FIGS. 2A and 2B show a transmitting portion and receiving portion, respectively, of a TDM OFDM transmission system in accordance with the principles of the present invention.

FIGS. 3A and 3B show an alternative embodiment of a TDM OFDM transmission system in accordance with the principles of the present invention.

FIG. 4 shows a transmitting portion and receiving portion, respectively, of a conventional OFDM transmission system.

FIG. 5 shows a general assignment of available subcarrier frequencies for use by three services in a conventional OFDM transmission.

FIG. 6 shows a specific conventional assignment of ten, twenty and thirty available subcarriers for use by three services, respectively.

FIGS. 7A and 7B show a relevant portion of a possible fading scenario in the conventional example shown in FIGS. 5 and 6 in which five of the subcarriers of the first service are hit by a frequency selective fade.

DETAILED DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

Instead of assigning respective portions of the available subcarriers in the frequency domain only to each requesting service as in the conventional techniques, the present invention provides a time division multiplexing (TDM) approach to dividing up the bandwidth of a channel among a plurality of requesting services. In accordance with the principles of the present invention, all available subcarriers for use by an orthogonal frequency division multiplex (OFDM) symbol are assigned to one particular service for a period of time, and then all available subcarriers are assigned for use by a second service for a second period of time, and then all available subcarriers are assigned for use by a third service for a third period of time, etc., until all services have been serviced. Then, all available subcarriers are again assigned to the first service, and the process repeats.

The amount of time during which each service has all available subcarriers assigned thereto relates to the respective bandwidth requirements. For instance, if all services require equal bandwidth, then all services may have equal time allotments of all of the available subcarriers. In this case, one symbol may be transmitted by each service using all necessary subcarriers for a period of time, and so one for all services. On the other hand, if one service requires, e.g., twice as much bandwidth as the other services, then that service preferably has use of all necessary ones of the subcarriers for twice as long as the other services.

Thus, time allotments of all available subcarriers to each service, one (or more) at a time, provides a time division multiplexing feature above the orthogonal frequency division multiplexing of the OFDM transmission system to provide TDM OFDM in accordance with the principles of the present invention.

The TDM transmissions are preferably frequency frame independent. Thus, a single service may have access to all available subcarriers at any one time even though there are many more requesting services. After the first service utilizes all available subcarriers for as many symbols as are required per an established bandwidth for that service, the next service will then have access to all available subcarriers for as many symbols as are required by its bandwidth, and so on.

Any one service may utilize any number of the available subcarriers for any period of time. Thus, e.g., a service may use just one or two subcarriers to transmit only one symbol (low bandwidth requirements) or may use all available subcarriers a plurality of times to transmit a large number of symbols (large bandwidth requirements). Thus, different services may take respectively different numbers of subcarriers for different periods of time to transmit their respective data, according to their bandwidth needs.

A full 'round' of each requesting services' timed multiplexed transmission over the relevant channel is referred to herein as a superframe. Superframes containing TDM OFDM symbols from a plurality of services in accordance with the principles of the present invention may be variable in length. The length of the superframe depends upon the length of one cycle of transmissions from all requesting services.

Furthermore, as new services are added or deleted, the length of the superframe preferably changes in length accordingly. An example superframe **800** is shown in FIG. 1.

FIG. 1 shows a plurality of full data frames **802–810** each corresponding to one use of all available subcarriers, and a

portion of a last data frame 812 comprising a single superframe 800. In FIG. 1, although it might appear to be more practical and simpler for implementation purposes for each service to take up a whole integer number of frames, this may be wasteful particularly if the frame size is not proportional to the exact bandwidth requirements of the requesting services. Thus, the present invention contemplates and relates equally to a non-frame length based TDM OFDM system wherein the TDM boundaries or 'slots'0 for each service in each superframe 800 do not necessarily lie on data frame boundaries defined by single uses of all available subcarriers, e.g., as depicted by the data frames in the frequency domain 802–812 shown in FIG. 1.

Based on particular application requirements, a first service 1040 is assigned all available subcarriers to transmit a first OFDM symbol (or symbols) 900. Note that the example of FIG. 1 shows that the OFDM symbols(s) 900 of the first service 1040 require the time otherwise required by a first conventional data frame 802 and a small portion of a second conventional data frame 804. The present invention is applicable to the use of less than the time corresponding to a single conventional data frame for the transmission of symbols of a single service of more time than a single conventional data frame, or exactly the same amount of time as a conventional data frame.

After the first service 1040 has had access to all subcarriers for its allotted TDM time necessary to transmit the number of symbols to achieve the desired throughput, the second service 1060 is then assigned access to all available subcarriers for transmission of, e.g., OFDM symbols 902 and 904. Note that in the disclosed example the second service 1060 is assigned most of the time corresponding to a second conventional data frame 804 and most of the time for a third conventional data frame 806, but not the full time for either the second or third conventional data frames 804, 806.

A third service 1080 is assigned access as necessary to all available subcarriers for a portion of time reflective of the required bandwidth for the third service 1080. In the disclosed example, the third service 1080 requires a larger bandwidth than the second service 1060, which requires a larger bandwidth than the first service 1040. The transmitted OFDM symbols 906–910 of the third service 1080 require, in the disclosed example, all of the time corresponding to the length of a conventional fourth data frame 808, all of the time corresponding to the length of a conventional fifth data frame 810, and a majority of the time corresponding to the length of a conventional sixth data frame 812.

In accordance with the principles of the present invention, services need not transmit in every superframe. For instance, depending in particular on the bandwidth requirements of the specific service, the services may utilize a variable number of the available subcarriers based on a variable number of OFDM symbols on a superframe to superframe basis. Moreover, this variation in the number of the available subcarriers used by each service on a superframe to superframe basis may vary, e.g., on a periodic basis or on a random basis.

FIGS. 2A and 2B show exemplary TDM OFDM transmission and receiver systems, 120, 130, respectively, in accordance with the principles of the present invention.

In particular, the transmission system 120 includes a bits to symbols mapper 402 and an OFDM transmitter and radio frequency (RF) front end 404 which are otherwise as in conventional systems. However, between the bits to symbols mapper 402 and the OFDM transmitter and RF front end 404 the TDM OFDM transmission system 120 includes a TDM symbol rearranger 150.

The TDM symbol rearranger 150 receives a serial bit stream containing data relating to a plurality of services and rearranges the bit data to achieve TDM timing, e.g., as shown in FIG. 1.

For instance, the TDM symbol rearranger 150 rearranges the data such that the data in the bit stream from the bits to symbols mapper 402 and corresponding to the first service is presented first to the OFDM transmitter and RF front end 404 corresponding to the length of time that all subcarriers are assigned to the first service. Thereafter, the data corresponding to the second service is presented to the OFDM transmitter and RF front end 404, and then the data corresponding to the third service is presented to the OFDM transmitter and RF front end 404.

A buffer may be included between the bits to symbols mapper 402 and the TDM symbol rearranger 150 as necessary.

The TDM OFDM receiver system 130 shown in FIG. 2B contains opposing modules as to the TDM OFDM transmission system 120 shown in FIG. 2A. For instance, the TDM OFDM receiver system 130 includes an RF front end and OFDM receiver 406, a TDM symbol arranger 160 (and buffer as necessary), and a symbols to bits demapper 408. The individual RF front end and OFDM receiver 406 and the symbols to bits demapper 408 components correspond substantially to those elements found in a conventional system, e.g., as shown in FIG. 4. However, the inventive system performs TDM multiplexing in the TDM symbol arranger 160.

FIGS. 3A and 3B show an alternative embodiment of a TDM OFDM transmission and receiver system in accordance with the principles of the present invention wherein instead of a serial stream of bits including data corresponding to a plurality of services, parallel access to the data regarding each of the plurality of services 1040–1080 is provided. In such a case, rather than rearranging a serial stream of bits, a TDM multiplexer 308 (or TDM demultiplexer 354 in FIG. 3B) merely time multiplexes (or demultiplexes) each of the data streams 1040–1080 for presentation to the OFDM transmitter and RF front end 310 (in FIG. 3A) and from the RF front end and OFDM receiver 352 (in FIG. 3B). The multiplexer 308 and demultiplexer 354 are operated in correspondence with the respective times that all subcarriers are assigned for access by the respective service.

In accordance with the principles of the present invention, use of all available OFDM subcarriers by each service lowers the relative percentage of subcarriers affected by environmental conditions, e.g., by selective fading, and thus increases the chances that an interleaver and/or FEC scheme relevant to that particular service will be capable of recovering from the lower percentage loss for that sequence of symbols. To this end, it is preferably that each requesting service use as many subcarriers as possible to reduce the likelihood of a high percentage loss of data due to selective fading.

While the invention has been described with reference to the exemplary embodiments thereof, those skilled in the art will be able to make various modifications to the described embodiments of the invention without departing from the true spirit and scope of the invention.

What is claimed is:

1. A time division multiplexed orthogonal frequency division multiplexed transmitter, comprising:

a bits to symbols mapper;

a time division multiplex symbol rearranger in communication with said bits to symbols mapper; and

an OFDM transmitter and radio frequency front end in communication with said time division multiplex symbol rearranger.

2. The time division multiplexed orthogonal frequency division multiplexed transmitter according to claim 1, wherein:

said time division multiplex symbol rearranger is adapted to rearrange an input data stream to provide output symbols relating to one requesting service at a time.

3. The time division multiplexed orthogonal frequency division multiplexed transmitter according to claim 1, wherein:

said time division multiplex symbol rearranger includes a multiplexer to multiplex any of a plurality of symbol streams to said OFDM transmitter and RF front end.

4. The time division multiplexed orthogonal frequency division multiplexed transmitter according to claim 1, wherein:

said time division multiplex symbol rearranger is adapted to output symbols relating to a single service for transmission by said OFDM transmitter and radio frequency front end using substantially all available subcarrier frequencies relating to a particular channel.

5. A time division multiplexed orthogonal frequency division multiplexed receiver, comprising:

a radio frequency front end and OFDM receiver;

a time division multiplex symbol arranger in communication with said radio frequency front end and OFDM receiver; and

a symbols to bits demapper in communication with said time division multiplex symbol arranger.

6. The time division multiplexed orthogonal frequency division multiplexed receiver according to claim 5, wherein:

said time division multiplex symbol arranger is adapted to arrange an input data stream to provide output symbols relating to one requesting service at a time.

7. The time division multiplexed orthogonal frequency division multiplexed receiver according to claim 5, wherein:

said time division multiplex symbol arranger includes a demultiplexer to demultiplex a symbol stream to said symbols to bits demapper.

* * * * *